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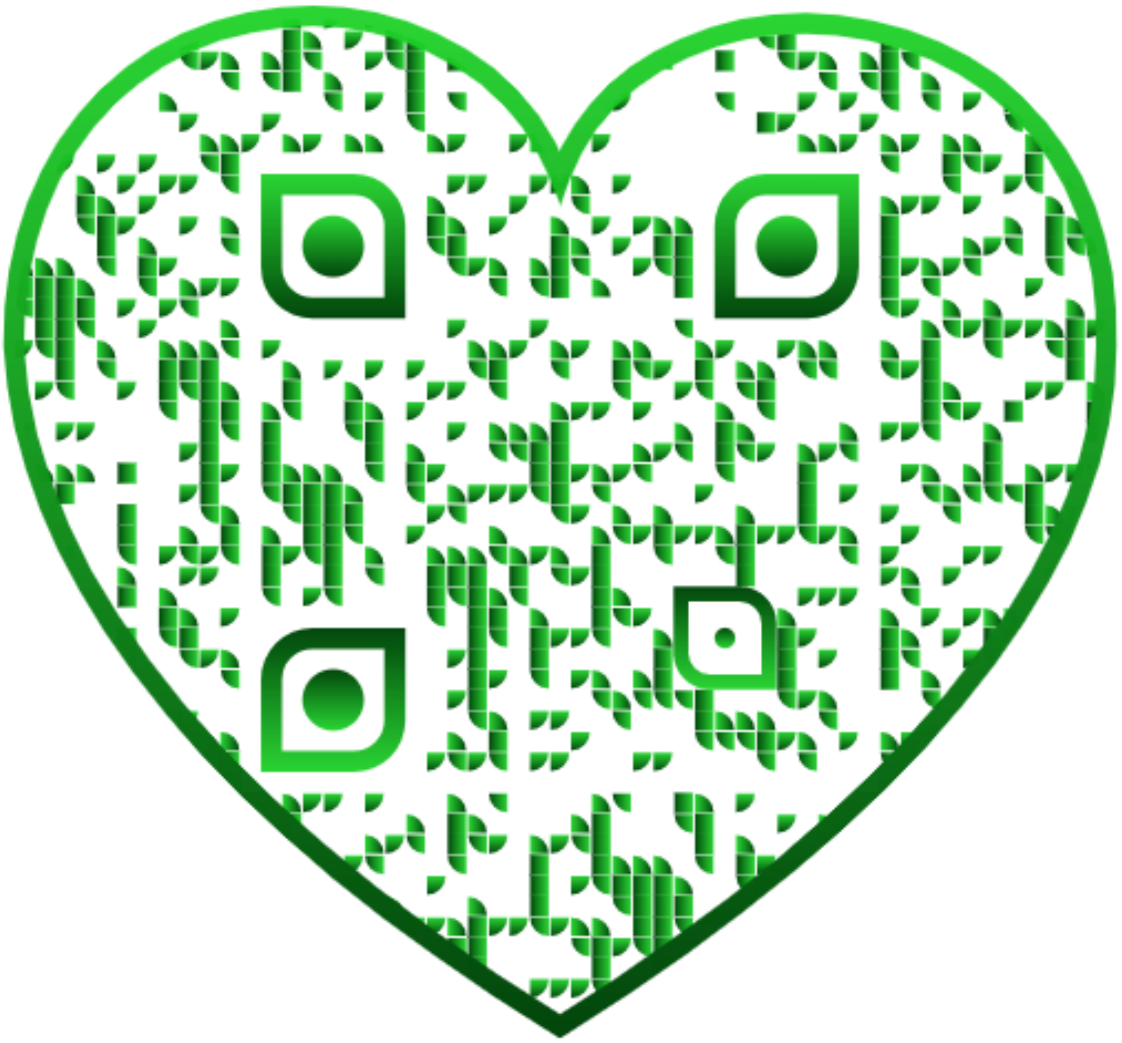

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TEMPTING FATE

There's something inspiring about stories of unexpected success. Back in the day, an influential DJ might take a fancy to some obscure B-side, and a three-year-old flop would become a hit single. Today, an impromptu TikTok video can spark a craze for shanty singing. Or, closer to home, a niche microphone familiar only to broadcasters and studio engineers suddenly becomes the must-have choice of podcasters, live streamers and gamers.

What makes these stories so compelling? For one thing, they hold open the idea that there's hope for all of us. No matter how unfashionable our genre, or how small our audience, maybe — just maybe — fate will find a path to stardom. But even for those of us who don't harbour the ambition to be a star, there's something peculiarly satisfying about this kind of off-piste success.

So often, in the music industry and beyond, it feels as though the world works to a script. Artists are manufactured by executives whose spreadsheets tell them that a precisely calibrated blend of Ed Sheeran, Nina Simone and the Red Hot Chili Peppers is what today's listeners will find irresistible. Products are cooked up by marketing departments who insist that they need something that's yellow, six inches long and can be manufactured at 99p a pop.

When raw talent breaks through from nowhere, or an unglamorous, engineering-led product becomes a hit, it provides a welcome corrective to that cynical world-view. In this connected world, we're all painfully aware of how little influence we have over the course of events. It's good to be reminded that whatever shadowy corporate powers might be out there don't have it all their own way, either.

Most importantly, though, I think these stories demonstrate the importance of always trying to do our best. As songwriters, we shouldn't settle for boring lyrics or predictable chord sequences just because we're not writing for Adele. As producers and engineers, we should aim to make every artist sound as good as they can sound, even if they're currently unknown, or if we ourselves can't see the potential in them.

When Shure's engineers created the SM7, they weren't aiming for 'good enough'. They meant to create the best mic they could with the resources at hand. And, 50 years later, their decisions paid off in ways they couldn't possibly have predicted.

Quality doesn't always win out over the innumerable other factors that shape our careers or those of the products we use. But by ensuring the quality is there, we can at least open up the possibility of success.

Sam Inglis
Editor In Chief

"By ensuring the quality is there, we can at least open up the possibility of success."

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Printed in the USA
Not for re-sale outside North America

ISSN 1473-5326
A Member of the SOS Publications Group

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Arturia Acid V

Software Synthesizer

DAVE CRANMER

Acid V is a software emulation of the ever popular Roland TB-303. With all of the important original features dutifully included, new sequencing, modulation and audio capabilities have also been added.

You may have already noticed the addition of several extra knobs to the front panel. These are for distortion, which makes it easy to jump right into high-energy screaming acid leads without having to search for a non-flat 9V battery for that orange Boss pedal; a sub-oscillator with octave shift and a choice of waveforms; and a vibrato section with depth and speed controls. Vibrato can be assigned to notes in the same way as the traditional accent and glide. Gone, though, is the infamously difficult sequencer, replaced with a grid system that makes the note entering and editing process very easy and offers sequences of up to 64 steps.

Arturia have another classic on their long list of software synths: a special little squelchy box.

Sequencer & Arpeggiator

The interface is nice and clear throughout, although I do hope that with repeated use the silver paint wears off around the cutoff knob as with the original. I was slightly unsure of the glowing green rings around the knobs at first, but they provide a good visual feedback when connected to the various modulation sources.

One slightly bothersome detail is that the 'Advanced' button is used frequently to flip between the front-panel view and the sequencer view, but it's also the tiniest button, far, far away in the upper-right corner of the window. It would be useful if there was a keyboard shortcut to flip from one mode to another. One for an update, perhaps?

After locating and pressing the Advanced button you're presented with tabbed pages for Sequencer, Modulation and Effects. More on the Modulation and Effects in a moment, but first let's turn to the original 303's greatest flaw (or greatest asset, depending to whom you speak) — the Sequencer.

Acid V's grid sequencer works much as you'd expect, but has some interesting features including 'Polymetric' mode. This allows the number of steps before the end of the loop to be set independently for octave select, notes, slide, accent and vibrato. It's effectively independent control of the 'last step' position for each of those parameters, so that you can set a sequence going where the notes to which slide, for example, is applied drifts over time, to create interesting variations.

The sequencer can be set to adhere to a number of different scales, and it's possible to enter custom scales (with probabilities set for each note). There is a set of traditional scales, and a set that's geared specifically towards acid melodies. There's also the option to fill in the blanks with a pseudo-random note generator.

The Transmutation section has a pleasing Erlenmeyer flask graphic, which can be used to fill in the blanks to a preselected scale. The range between lowest and highest note is set by the amount of liquid in the flask, and there is also control of Density, ie. the total number of notes that appear in the sequence.

Hover above the octave, slide, accent and other labels and you will get dice icons for further randomisation. Click and drag each die up or down to increase or decrease the likeliness of that parameter being applied to the steps. Dragging a die down to zero is also a quick way to clear the settings on a particular parameter row without having to use the global erase tool. No problem if you erase something accidentally, as an extensive undo history is provided.

There is also a comprehensive arpeggiator, which adds a whole dimension of really fun live pattern generation. It took me approximately five seconds to get this set up with a MIDI keyboard and I spent a lot of time just rocking out playing arpeggiated chords with one hand and knob twiddling with the other. It's great fun to get the arpeggiator going and then randomise octave jumps and slides, and Polymetric row settings are allowed in this mode as well as in sequencing mode.

Extras

Movement is everything with a 303 — after all until Phuture started twiddling the knobs it was just a failed bass accompaniment box. Now it's one of the worlds best-known synths, and people are even making software copies of it! With this idea of movement in mind, Acid V has three comprehensive modulation sources that can be linked to all of the front-panel knobs (but not selector switches). Simply drag and drop the cross arrow icon that appears when you hover over the animated modulation graphic within the tab. You may »

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Clicking the Advanced button reveals tabs for the sequencer, modulation and effects pages.

» assign two modulation sources to any of the knobs that have a glowing green surround. Double-click to disconnect the modulation currently assigned to the knob.

Regarding Phuture, I'm sure a lot of you will be thinking, 'Ah, but MC Turnip Centrifuge got there earlier!' Just for the record, check the date on Charanjit Singh's *Synthesizing: Ten Ragas To A Disco Beat*, and you'll realise he must've picked up his 303 directly from the factory on the day it was released, and started recording during the bus journey home. His album consists of acid tracks from 1982, long before the genre had a name. Admittedly the album is a bit like the Madagascan tenrec, which looks very much like a hedgehog but evolved quite separately.

The three modulation sources can be set as envelopes or LFOs and triggered by key press, note, accent or sequencer restart. There are buttons to select basic modulation shapes and a preset menu for more complex ones, many of which work better as envelopes than LFOs, and you can also draw your own shapes.

The last tab in Advanced mode presents a choice of 17 different effects (in addition to the choice of distortion types on the front panel), with the ability to load a bank of four at any one time. Personally I think the perfect TB-303 solo doesn't need a great deal of long ping-pong delay, but there's no harm in having the option, and the effects can of course be used subtly. The list of effects includes reverb, delays, compressors, filters, distortion, a bit-crusher, chorus, flangers, phaser and stereo pan. The effects interface is clear and easy to use, with informative graphics, and doesn't get in the way of using the rest of the 303. The whole hierarchy of Acid V has in fact been well planned — you can get started quickly but delve into each further level of detail as you wish.

The front panel has a button just above the Acid V logo that activates 'Trimmers Mode'. When activated a new set of controls pop up, giving the impression the casing has been dismantled to expose the inner electronics. I've never seen electrolytic capacitors that size or colour, but we can forgive the interface designers as it's a nice way of presenting the additional parameters for tweaking, and emphasising that they are non-standard.

Within this inner electronic area there are little trimmer potentiometers for adjusting a few things including bass boost, the cutoff range and accent attack time, and pitch tracking can also be modified. Ed DMX of DMX Krew has mentioned that his 303 is quite out of tune, with high notes being sharp and lower notes being flatter than they should be, but he likes it like that. So adjust pitch tracking for bonus analogue wonkiness.

In Use

Arturia have faithfully recreated the sound of the 303, including the odd thing that it does where turning the resonance right down creates a disproportionately high increase in volume. Note that although the master volume control appears to be missing from the front panel, there is a little dial for this in the top-right corner. Useful if you want to compensate for the drop in volume when the resonance is turned up.

I set up an identical pattern in Acid V and on the original TB-303, with no effects, and compared the two side by side. The most noticeable detail whilst attempting to match the two was that the tuning of the original 303 seemed to drift very slightly every time it was restarted. It was also easier to fine-tune the pitch in software — left-clicking pitch changes in semitones and right-clicking any knob gives you fine tuning. The sounds were otherwise very, very similar, certainly within the tolerance of early '80s analogue components. The cutoff on the original was very slightly more 'whistley' when turned fully clockwise, but it's likely that another 303 would sound different again. I couldn't find any faults when comparing the two, and anyway perhaps we ought to stop these comparisons and just get on with making some music. Incidentally, simultaneously running the same pattern in Acid V and the hardware 303 with slightly differing tempos immediately created a track that I call 'Steve Reich Acid'.

Connecting a BeatStep Pro and assigning knobs in MIDI Learn mode was very straightforward, and definitely recommended for 'head back with eyes closed' in-the-zone filter manipulation. I think potentiometers with physical stops would be better than the rotary encoders I had to hand, but pretty much anything has to be better than the fiddly little knobs on the original. Such an amazing huge sound in such a terrible plastic box!

Conclusion

Acid V is an excellent TB-303 clone with a good selection of additional features that encourage exploration. Arturia have cleverly not strayed too far from the original silver box, keeping additions like vibrato relatively subtle rather than attempting to turn it into a modular synth. This is definitely about working on cool patterns and then tweaking the sounds, with the live arpeggiator being particularly fun. Another charismatic addition to Arturia's soft-synth arsenal.

summary

An accurate and versatile TB-303 soft synth, with plenty of extras and at a tiny fraction of the cost of an original.

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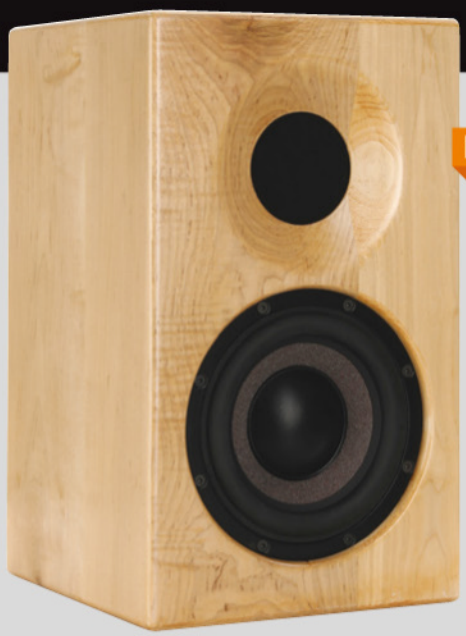

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CHRIS KORFF

Origin Effects may be familiar to you as the company who had the inspired idea to build an 1176 into a guitar pedal. They've since come out with a range of top-quality (and commensurately priced) stompboxes, and their latest models cater to bass players, with the Bass Rig Super Vintage promising the sound of an Ampeg SVT, and the Bass Rig '64 Black Panel being based around Fender's Showman and Bassman models.

The two are nearly identical in terms of I/O and features. Both sport a single instrument input, a DI output on XLR, and an amp out jack. The DI out incorporates an analogue speaker simulator, while the amp out is taken directly after the preamp stage, and includes some EQ options to allow it to work optimally with either a traditional bass amp or a full-range backline setup.

Knobs labelled Output, Blend (wet/dry mix), Drive, Bass, Mid and Treble do exactly as you'd expect, with the EQ sitting before the overdrive and thus giving you a huge range of tones to play with. The Amp Out EQ section applies only to the jack output, and includes HF and LF trim pots, as well as a three-way 'Horn Cut' toggle switch, which allows you to high-pass filter the amp output at either 2 or 4 kHz (or not at all, if you're using a traditional cabinet). Also common to both pedals is a further pair of mini-toggles, one of which engages a ground lift while the other concerns the behaviour of the on/off footswitch. The latter's FX and FX+Byp options decide whether the speaker sim stays active when you bypass the pedal; in FX mode the whole pedal is bypassed, while in FX+Byp, you keep the speaker simulation when you deactivate the pedal, which is handy if you want to use it as a distortion effect.

Where the Super Vintage and '64 differ, in terms of controls, is the two toggles at the top of each. On the former, the upper-left switch sets the EQ's mid frequency to 220Hz, 800Hz or 3kHz, while on the '64 it selects between Black Panel ("a controlled, scooped tone") and Tweed (higher-gain, more mids) circuitry. The upper-right switch on the Super Vintage can engage either a top boost (post preamp) or a low-cut filter (pre-preamp), while the equivalent on the '64 either adds a drive-dependent top boost (whose effect reduces as you increase the amount of drive), or, in the Fat setting, "reconfigures the circuit from Showman to Bassman spec".

Both pedals require a 9V DC 150mA power supply, of the usual centre-negative variety. No PSU is provided.

In Use

So, how do they sound? In a word, excellent. I tried the Super Vintage first, and before long, I'd forgotten about reviewing the thing and was playing for the pure joy of it — I can honestly say I haven't enjoyed noodling around on bass that much in a long time. I'm not an amp snob, and when I gig I often forgo an amp altogether for the sake of convenience, so I'm very used to the sound of a straight DI. The lovely warm tone I was hearing from the Super Vintage was nothing less than inspiring, and if I had one I'd think nothing of ditching my amp altogether.

The EQ controls have a wide enough range to satisfy most demands, with the mid frequency selector allowing you to scoop or fill out the sound, or notch out mechanical noises, with ease. The Drive knob behaves predictably and progresses smoothly through all of its travel, but it's in the interaction between the EQ and Drive controls that the magic lies. The Bass and Mid controls, in particular, can have a huge effect when you're being liberal with the dirt, with the EQ coaxing some lovely harmonics out of the ensuing Drive circuitry. There really is a world of tones available here, and all of them are lovely.

It was a similar story when I tried the '64 Black Panel, except that the onset of overdrive was rather more abrupt than on the Super Vintage, so I had to be a little more careful when setting the EQ and Drive controls. That said, at higher gain settings I was able to achieve some wonderfully raucous and metallic sounds that I couldn't on the Super Vintage, and which would happily grace any punk record.

As someone who uses roundwounds, and measures the lifespan of bass strings in years rather than weeks, my preferred tone is very much on the warmer end of the spectrum, so for me the Super Vintage was absolutely ideal. If you prefer more bite to your bass, you'd do well to check out the '64 Black Panel. Both, though, are fabulous, and will serve you very well indeed in both live and recording scenarios.

summary

Two excellent, responsive and lovely-sounding 'amp in a pedal' stompboxes. Both offer convincing amp sounds, but veer in different sonic directions when you crank up the gain. For cuddly overdrive, the Super Vintage is excellent. For more gnarly distortion, the '64 Black Panel might suit you better.

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MATT HOUGHTON

It's been many years since a genuinely 'bad' mic preamp made its way into my studio, and those built into most interfaces do a pretty good job of amplifying what the mic is picking up — so any standalone preamp released today had better do something more than boost a mic signal! Engineers are often drawn to vintage preamps (or modern ones based upon them) specifically because of the character they impart, be that the relatively subtle effect of transformers, discrete opamps or valves in normal use, or the distortion as you start to 'drive' them with hotter signals. I find it refreshing when a manufacturer does something more than clone a colourful classic, though, and the Dark Storm, manufactured by US-based Lightning Boy Audio, ticks that box.

Into The Storm

An original Class-A design that can deliver up to 75dB of gain in total, the Dark Storm features both mic and instrument preamps, and is based around a custom JFET opamp and a number of transformers. The mic input is accessed via an XLR on the rear of the 1U half-rack case, and the input is balanced using the company's own LBA-MCX4 mic input transformer. The instrument input is unbalanced and its connector is a front-panel quarter-inch TS jack. Its DI circuit employs a JFET and there are two transformers in the signal path — an LBA-MC15 and the aforementioned LBA-MCX4. (If interested you can find the specs of all these transformers on the company's website).

On the rear are separate ground lift switches for each input. Power comes in on a 2.5mm centre-positive connector from an external 48V DC 2A adaptor, which terminates in an IEC inlet, for connection to mains AC. The Dark Storm can pass power on to up to seven more units courtesy of a daisy-chain output, helping to keep the cable mess to a minimum. Each unit's power circuit creates the ±18V power rails for the audio circuitry, and is internally isolated from any others in the chain.

The main +4dBU 600Ω line output has an XLR connector, and is balanced using one of the company's chunky Fat Sally range of transformers. But you can also send an unbalanced high-impedance (145kΩ) signal out, simultaneously, through a quarter-inch TS jack, with the impedance conversion performed by another of the company's transformers, this time an LBA-20SX steel-core model. You could, for example, use this output for true zero-latency monitoring of the input signal while passing the balanced feed to your interface/recorder, or to route your mic signal out to a guitar amp or guitar pedals, while recording the balanced output dry. Or, guitarists might want to use the Dark Storm as a preamp and feed the unbalanced output into the power stage of their amp... In short, it's a nice feature for the experimenters among us, and increases this preamp's versatility.

In terms of user controls, there are two pentagonal pointer knobs, emblazoned with a lightning bolt (part of the company's logo). The left one is the main gain control, and right one is labelled Trim. Both are continuous pots and have scale markings for recall but no numbers. There's no metering either, but this is all by design: the manufacturers want to encourage you to listen and set things accordingly. Gain does what it says on the tin, while Trim is a passive attenuator. This isn't simply about precision, though; the two act at different points in the signal path, with Trim coming just before the output transformer. This configuration makes the Dark Storm as effective as a saturation or distortion processor as it is as a preamp.

There are also five backlit buttons. One (red) switches the device on/off (it's nice to see this on the front, where users actually want it!). The others are blue, and include the usual polarity invert, 48V phantom power and (58Hz) high-pass filter switches. Then there's the intriguingly named Old button. As the name implies, this changes the sonic character somewhat, and with a high- and low-end roll-off, a slight thickening effect, a change of impedance (from 1.4kΩ to a lowly 150Ω; the DI input remains at 1MΩ), and a chunky level boost, this is almost certainly accomplished using a transformer.

Storming Performance

For a box with so few user controls, the Dark Storm offers impressive tonal versatility, and with both the mic and instrument inputs »

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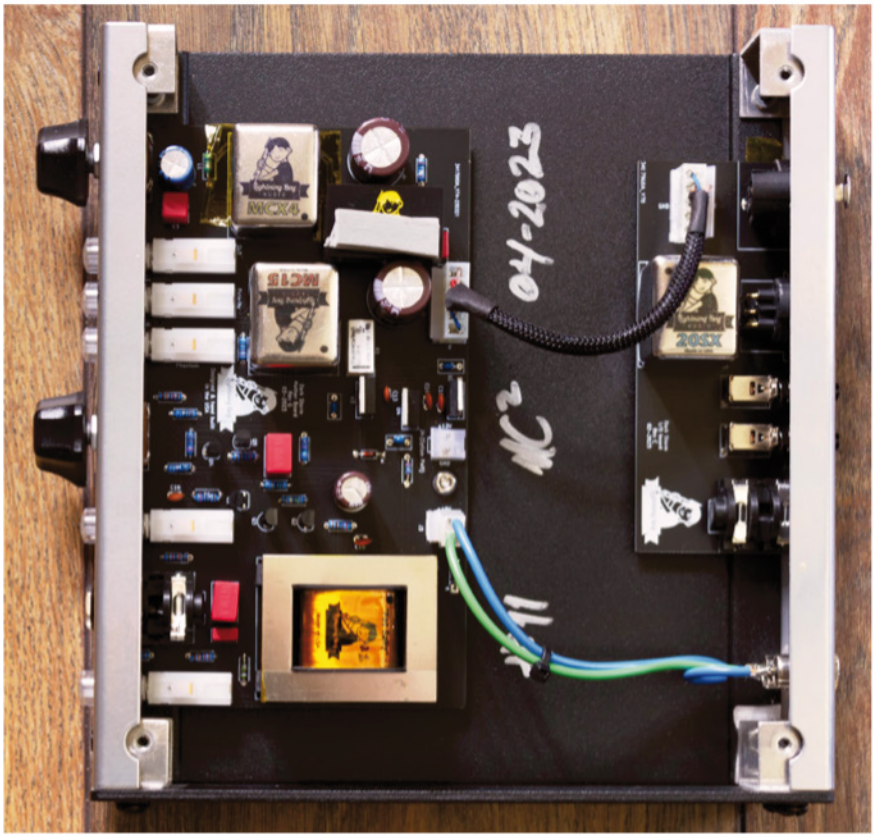
The Dark Storm offers both a +4dBu transformer-balanced line output and an unbalanced high-impedance output, both of which can be used simultaneously.

» feeding the same circuit, operation is trivially easy. Just turn down the gain, plug in your mic or instrument (switching phantom on if required), set Trim to a roughly 12-o'clock position and turn up the gain to get the signal to the desired level. From there, you can just back off gain and turn up Trim if you want a cleaner sound, or back off Trim and turn up gain for more character. And in the final stages of the gain knob's travel, you're greeted with some real attitude.

So simple, but so addictive. And what character there is on offer here! As part of my review tests, I tracked overdubs for a whole pop/rock song, including using it on a mono drum room mic, and I was blown away by the ease and immediacy with which I could dial in colour to taste. On the 'clean' end of the gain control, the sound is not clean exactly, but cleanish, and slightly snappier and more 'lively' somehow than many. At the other end, particularly those final three 'coloured in' lines, the effect is not subtle. Think overdriven console channel, but a pleasing sort of breakup, without so much of the 'fizz' that some preamps exhibit when cranked. If you experiment with the gain structure you can find more nuance too, not least by manipulating the level being fed into the output transformer. And this is all before you throw the Old switch into the equation. This is not something you can easily A/B compare, since the gain change is significant, but it does plenty to shape the tone, and to tame the distortion if you push things that far. A slightly tube-like quality, if you like, and I also found that, presumably due to the input impedance change, it could be useful for coaxing a different tonality out of my dynamic mics when used on a guitar cab.



A peek inside: Lightning Boy have based the Dark Storm around a discrete JFET opamp and a number of their own custom-wound transformers.

Rain Supreme?

The Dark Storm *can* do relatively subtle things, but this is clearly not its reason for being, and it's not likely to be your first choice for recording classical flute or piano. For anything else, though, where you want the sound to seem that bit larger than life, it works very well. It has the range to do pleasing things on pretty much anything and, importantly, different things for a range of sources in the same session. From close-miked, breathy, whisper vocals through my C414 to raunchy rock vox through an SM57. From a thick, rounded DI bass guitar to a distorting electric guitar preamp or a box to breathe new life and dimension into synth parts... You get the idea. Just a few controls can do so much. It's one of those toys that awakens the creative urge, and encourages you to play and experiment — and when I'm in that mood I find it a whole lot easier to make music!

summary

So much more than 'just' a mic preamp, the Dark Storm is as much about colour and character as it is about gain. It practically begs you to play and experiment!

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Sonuscore Fragments: Modern Percussion

Sample Library

Sonuscore combine organic percussion sounds with a creative sequencing engine.

JOHN WALDEN

While Sonuscore's latest release — Fragments: Modern Percussion — will undoubtedly appeal to their core media composer fanbase, I suspect this Kontakt-based instrument might also be of interest to a wider audience. Fragments features a three-lane step-sequencer sound engine. However, the lane labels — Bass Drum, Snare Drum and Fragments — have been chosen to imply a role rather than a specific sound, because the compact (the whole library is only 100MB in size) collection is built from more organic-style sound sources. Yes, the Bass Drum and Snare Drum lanes might provide the same low-end and punchy low mids that a conventional kick and snare might, while the Fragments lane then takes on the high-end role of the hi-hat but, in each case, more organic sounds fulfil the task.

Thanks to some very clever elements within the playback engine, you can coax a pretty diverse range of sounds from the underlying samples. All the key features are spread across three tabbed pages; Main, FX and User. The Main page is where the bulk of the action is at. As well as access to the tag-based preset browser, this is where you get access to the key settings for each of the three sound slots. For example, at the top of each slot you can activate any combination of the five available sound types — Wood, Metal, Mech, Body and User — providing a simple means to customise the sound generated. The User slot allows you to add a single sample of your own to each of the three sound slots, providing further flexibility. These user samples can be dragged and dropped into the User page prior to use, where you also get options to adjust the attack and volume of the sample.

Towards the base of the Main page are the three step sequencers. These offer individual step counts (any value up to 32 steps) and step lengths (with dotted and triplet options available) providing plenty of rhythmic flexibility. With the Engine button engaged (the default), trigger keys for all three sequencers together, or each individual sequencer, are mapped on C3 to F3. However, if you disengage the Engine, the key mapping is adjusted, letting you play all the samples manually.

Wait, There's More

Some of the further options are particularly interesting. First, the three Sound knobs allow you to change which of the underlying samples is used within each sound slot, while the small slider underneath the knob adds a degree of (rather wonderful) randomness to the sound selection during playback. Second, engaging one (or more) of the Density buttons lets the sequencer add ghost notes to any unused step in the respective sequencer and, by default, you can control the level/velocity of these ghost notes via the mod wheel (CC1). Third, the two dice icons above each sequencer allow you to both generate new step sequences (by rearranging the current pattern) or, on playback, to generate a rearranged pattern each time the sequencer loops. This latter option is particularly effective when used with the Fragments slot, but all the dice button options are great for generating new pattern ideas. Fourth, while your host is running, the mini keyboard icon located bottom-left provides drag-and-drop MIDI export to your DAW for editing and playback with the Engine disengaged; very neat.

The User page includes options for configuring a selection of sound modulation options (such as the mod wheel density behaviour mentioned above) and these include control over the individual filters found within the FX page alongside other effects such as a global reverb and compressor, and individual delay and high/low-cut controls for each sound slot.

Conclusion

The sum of all these parts ends up being a really impressive — and very creative — percussion tool. It's perfectly capable of providing a full percussion performance but, equally, also works great alongside a more conventional set of drum/percussion sounds. Fragments' sound set is described as 'organic' and that feels about right — it's quirky but very cool — and, as such would work brilliantly in a contemporary music-to-picture context or in pop or electronica production. As so often, Sonuscore have managed to combine a carefully targeted sample base, some creative sequencing tools, and clever sound-design options to make a very creative instrument that's also a pleasure to use. Fragments: Modern Percussion is perhaps something of a specialist product, but it's also a mighty fine one and competitively priced to boot. ■■■

summary

Combining some quirky percussion sounds and a powerful pattern sequencing engine, Fragments is wonderfully creative.

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Acon Digital Remix

Source Separation Plug-in

There are now some impressive offline source separation tools — but this one works in real time!

SAM INGLIS

Source separation sometimes seems like a technology in search of an application. It's remarkable that we can now unbake the audio cake, extracting individual sounds from within a mixed recording — but how useful is this in a real-world music-production context? The results can be impressive, but they're never completely free of artefacts so, where they're available, it's always better to go back to the original multitracks. And where the multitracks aren't available, that's often for a very good reason!

Such applications as there are have also perhaps been restricted by existing implementations of the technology. Apart from AudioSourceRe's RePan, none of the tools I've tried has operated in real time, and many require audio to be uploaded to a server before it can be separated. Moreover, user interface design hasn't always made it easy to fully exploit the possibilities. So I was intrigued by Acon Digital's new Remix, which operates as a real-time plug-in in AAX Native, VST and AU formats, and which promises an extremely simple, streamlined interface.

Both promises are undeniably fulfilled, and once you instantiate Remix on a track or bus, you'll be greeted with a freely resizable mixer window containing five channels labelled Vocals, Piano, Bass, Drums and Other. Each has the usual fader, solo and mute controls, plus a Sensitivity dial that runs from -100 to 100 percent. This, in essence, controls the 'weighting' of the detection process; the more you turn up the Sensitivity of the Vocals, say, the more likely it is that audio within the track will be interpreted as belonging to that source. If you leave all the controls in their default positions, the separated sources sum perfectly to recreate the original audio file. And in hosts that permit it, Remix can address multiple outputs, so you can split out the individual stems to different channels in the DAW mixer.

Stem Power

As luck would have it, the opportunity to review Remix arrived just as I finally ran across not one but two real-world uses for source separation. A friend sent me an old soundboard recording of a gig where the balance between drums, vocals and guitars was badly askew; and, having recorded a jazz trio live in a studio with no isolation booths, I found myself with uncomfortable levels of drum spill on the double bass mic. In both cases, I compared Remix with Hit'n'Mix's RipX, which probably achieves the most natural results of any source separation software I've tested to date.

I had expected Remix to be very CPU-intensive, but surprisingly, this doesn't appear to be the case. In fact, when running on an older Intel Mac Mini (not a particularly powerful computer) I could barely detect any system load at all in the Pro Tools System Usage window. However, it does introduce a processing delay of 11,264 samples, which is a quarter of a second at 44.1kHz, so it's not an option for live use.

It works, though. Everything in my test tracks was correctly identified, with the Other channel effectively becoming a guitar track, and Remix didn't falsely detect a piano where there was none. If you solo a channel, it's easy to hear the effect of the Sensitivity control; as you turn it up, the wanted source sounds clearer and brighter, but spill from the other sources begins to be introduced too. Remix was able to make a positive difference to both my test sources — but, perhaps inevitably, RipX delivered more natural results, with fewer obvious artefacts. I could go significantly further with rebalancing the live take using the RipX stems before warbling or watery noises began to be apparent, and on the double bass, RipX retained more upper-mid detail and articulation. This is hardly surprising when you consider that the offline ripping process took several minutes for a three-minute bass part: an offline splitter is simply able to throw more CPU cycles at the problem, and has no restriction on look-ahead time.

I think the real value of Remix, then, is that you can use it in situations where you couldn't or wouldn't use an offline alternative. There are plenty of times during sessions when you don't want to interrupt the flow in order to leave the DAW, set up an offline split and then reimport the results. Remix is highly affordable, delivers useful results and — uniquely — does so instantly. ■■■

summary

Supremely easy to use, Acon Digital's Remix has the unique ability to perform source separation in real time.

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Musik Hack Master Plan

Mastering Processor

MATT HOUGHTON

This great-sounding processor does more than make your music loud — but does it do enough?

At first glance, Musik Hack's debut plug-in Master Plan appears to be all about loudness: the biggest control is labelled Loud, and cranking it can certainly make things very loud. But that's only part of the story, because Master Plan also includes a host of useful simple, intuitive facilities to shape and refine the sound you feed into this limiter. The company suggest, in fact, that it offers enough control that you should consider throwing out whatever else you have on the stereo bus — I'd describe it more as a sort of stripped-down mastering chain, with a selection of well-judged tools!

Master Plan is available for Mac and Windows and supports AAX, AU and VST3 hosts. Twelve-month and perpetual licences are available, and authorisation to your computer is a simple online process; you don't seem to need to be online thereafter for it to function.

When it comes to my own mastering efforts, I tend to prefer using separate and fairly traditional processors. So, with its fixed signal flow and streamlined control set, Master Plan offers a very different approach than I'm used to, and it's fair to say that I approached this review with a little trepidation. But my apprehension turned out to be ill-founded. You *can* make things very loud with this plug-in, but as long as you use it with the requisite care and a discerning ear, you can do a lot to balance and sweeten things. The only real downside is that the minimal control set means there are quite a few things it *can't* do. Nonetheless, I was pleasantly surprised and genuinely impressed.

Exploring

By default, the GUI was pretty small on my 16-inch M1 MacBook Pro's screen (it maybe less so on lower-DPI monitors) and at that size the off-white text on a black background wasn't the easiest to read. But it's resizable, with five fixed settings ranging from XS to XL, and when set to XL it occupied about a quarter of my screen and was easy to read. You can also switch to a white background instead of the default black, which further improves legibility in my view. While we're on the GUI, many controls have a coloured 'glow' whose colour is cycled by default, apparently just for visual entertainment rather than to indicate something. Not being a 'spinner hubcaps' sort of person, I was pleased to see that this feature could be turned off in the settings window, and a static colour of virtual LEDs specified.

It's incredibly easy to get started, but you should first listen, as the default settings aren't completely transparent. Before you do anything, there's a gentle thickening and a presence boost (the latter probably as much due to harmonic distortion as EQ). Then, pick a preset, set Out (the output ceiling) to taste and crank the Loud knob, and you'll have instant gratification. You'll probably find that you can push the perceived loudness a bit further than on most limiters, and this seems to be achieved with some sort of pre-emphasis that compensates for the effects of heavy limiting. So far so ordinary. But handy Bypass (delay compensated) and Unity buttons allow you to compare the processed and unprocessed signals. If you engage Unity the perceived level remains the same whatever you do with the controls. That's great, since it allows you to make judgements based on the sound, not on the meter readings across the bottom. Push Loud up until you hear things start to break up too much and then back it off a little. Then disengage Unity and you'll probably have a grin on your face as you realise just how loud you made your mix.

Such a 'twist it and see' approach will only get you so far, of course, and for the best results you'll need to understand what the other facilities do and where in the signal path they do it — and while the GUI is largely intuitive, the control layout doesn't always reflect the signal flow. Top left, the In knob is an input gain control, and since there are threshold-dependent processors with no user-adjustable threshold, you're going to want to learn to use this to get the signal in the sweet spot, particularly if mastering mixes that have already had stereo bus processing applied. An adjacent clip LED warns you that the processor is saturating, and is an indication to back things off for a cleaner sound. Top right, the aforementioned Out knob sets the output ceiling anywhere from the default -0.1dB to -12dB True Peak. Next »

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The alternative white GUI.

to this are undo/redo buttons and the Bypass and Unity switches.

The main Loud knob increases the signal level going into what, to my ears, is a great-sounding combination of limiter and clipper. Up to 24dB of boost is available, and as the signal approaches the ceiling the algorithm does its thing. To the left of Loud are low (±6dB) and high (±8dB) shelf EQs. An 8dB boost might sound a lot for mastering, but the gentle slopes mean you can set this very high up the spectrum, with the EQ curve acting much more gently further down (an 8dB boost at 20kHz starts lifting somewhere in the 800-900 Hz region). Note that these EQ bands come early in the signal chain, so boosts and cuts here dictate what's fed into the limiting/clipping circuit, and if you need to rebalance things radically you may find it helpful to dial back the input level, to give yourself some headroom.

Below, a saturation processor called Thick can be set from zero to 100 percent — the plug-in already introduces harmonics and it was very rare that I wanted to use this additional thickener when mastering, but it could certainly be useful in shaping sounds other than full mixes. Adjacent is another EQ called Clean. This can be switched on/off, making comparison easy, and when engaged applies a dip of up to 2dB, centred around 200Hz. The idea is to allow you to clean up any of the mud or boxiness that can often build up in mixes in that region, and can become more obvious as you start compressing and limiting.

Further facilities include Multi, a multiband compressor which can deliver up to 6dB of gain reduction independently in three bands (the crossover filters are fixed). Additionally, there's Smooth, a gentle single-band compressor with fixed settings, while Tape is what the name suggests: an analogue-modelling tape simulation, again with mostly fixed settings. This models the head bump, HF roll-off and hysteresis, and offers a single slide control to apply the effect from zero to 100. A Calm button, again with zero to 100 slider, is designed to tame high-frequency harshness and this appears to be another cut-only EQ, delivering cuts at around 4 and 8 kHz, while a x2 button adds a similar pair of filters at around 6 and 12 kHz.

Across the bottom are numerical readouts of loudness, with LUFS-I, LUFS-S, True Peak and Crest Factor, all displayed simultaneously. In the Settings window, you can switch these readouts to change colour to indicate when you're hitting/exceeding loudness targets. It's a clever idea, though sadly these aren't user-definable targets — and it might have been nice to have one for the Crest Factor too. We also have what I think is an M-S based Wide control, which can make the mix wider but not narrower and sits at the start of the signal path. Finally, there's an array of monitor control facilities (mono, dim, a band-pass filter, and NS10 and phone emulations) to indicate how well the results will translate to consumer playback systems.

Right On Plan?

This all adds up to a really nice-sounding limiter/maximiser that can seem subjectively very clean and clear when pushing it really quite hard, and the Unity loudness compensation means you can make your judgements based on the sound and not just the numbers, which is as it should be. It also presents a range of options for cleaning up or thickening the sound. If, for example, you hear harshness, hardness or brittleness as you crank the Loud control, you can try countering it with a HF shelf, multiband compression, the tape emulation or the Calm EQ. At the same time, you have options to boost the low end with the tape sim, or with the shelf, or saturation... and have the ability to reduce any mud that such processing brings to prominence. And those prepared to dive a little deeper will find a pretty sophisticated signal chain with thresholds all sitting in the right place, allowing you to balance the Input and Loud knobs to obtain cleaner or dirtier results at or around the same loudness.

The Settings window allows you to change some of the aesthetics, including the meter behaviour.

Can it replace your entire mastering chain? It's certainly possible to do a lot with this one plug-in. It's not hard to coax wonderful results from it, but equally important is that it's hard to go very far wrong. For me, it's not quite the do-it-all plug-in that's claimed, and obviously not every mix needs to have character added. For mastering, I'm still going to want my M-S EQs, and my favourite compressors and saturators, and certainly a dither, the lack of which surprised me here. But I do love how this plug-in sounds, and what it can do (and do so conveniently), and the quality on offer does justify a not insignificant price tag — this is a well-judged signal chain, and I will definitely be making use of Master Plan going forward. ■■■

summary

An impressive-sounding limiter-centred mastering chain, Master Plan might lack the control of traditional tools, but is capable of great results — and helps to make bad ones unlikely!

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Digit Music CMPSR

MIDI Controller

Tired of keyboards? The Digit Music CMPSR might be the MIDI controller for you...

NICK ROTHWELL

Digit Music's CMPSR is, at first impression, a cross between a MIDI controller and a gaming device. It resembles a game controller, with a rather industrial-looking joystick and a bunch of RGB LEDs, but it's actually a class-compliant MIDI device which generates notes and chords. It's big — almost 20cm long — and its styling and control layout remind me of nothing so much as a 1980s mobile phone. Like a phone, it's primarily designed to be handheld, and in use will often require both hands. The underside is rounded and the device won't sit neatly on a table top, though there is an optional RAM Mount kit to bolt it to any convenient fixture. The battery-powered CMPSR sends MIDI via its USB-C port, which is also used for charging, or it can send via Bluetooth for truly mobile, cable-free performance.

How Does It Work?

The CMPSR is primarily a generator of notes, and the centre-weighted joystick is the playing interface. Its eight compass directions play, by default, one major scale octave from C to C. The joystick is velocity sensitive: the faster you whip it from centre to edge, the louder or brighter the note. Velocity sensing can be turned off in favour of fixed velocity, but there's no editing of response curves — you'll have to do that in your instrument or DAW.

The buttons are essentially for mode selection, and change colour to indicate current settings. One button is basically a sustain pedal: it toggles on and off. Another puts the device into chord mode, so that the joystick starts playing triads. These are mapped to the current scale, so from C you get Cmaj, Dmin, Emin and so on, rather than chromatic transpositions. The result is rather simplistic, but enough for rattling off convincing synth stabs at a techno gig. Two further buttons enhance the chord generation: one adds 'extensions', cycling through added seventh, ninth and fourth voicings, while another generates inversions. The bottom two buttons offer octave transposition or, when pressed together, activate a short-lived 'accidental' mode, which flattens all the notes in the next joystick gesture by one semitone, resetting after the note or chord is played. A bundled application makes this mode editing a bit easier.

In Use

These functions are all performative, and can be punched in and out whilst playing, which with a bit of practice leads to a complexity of performance that belies the CMPSR's simple voicing algorithms. There's a setup mode allowing a global switch into a minor key, and transposition of the root note (showing the letter name of the scale on the eight LEDs, which is helpful). A 'pro' firmware version lets the buttons transmit MIDI determined by colour-coded device 'pages', and activates various MIDI CC outputs from the joystick, turning it into a gestural controller.

After a while I did start to get a bit bored of triggering chords, but I dropped an arpeggiator into Bitwig Studio and that really brought the CMPSR to life, generating constant variations on Philip Glass as I walked up and down the scale and switched chord voicings on the fly. The joystick controller mode was welcome as well, although controller and note messages are issued at the same time, which is rather unavoidable with only one joystick. If you don't mind that restriction, there's potential in what the CMPSR can generate, especially with a bit of clever DAW programming.

summary

The CMPSR is a game controller-like MIDI device for triggering notes and chords with a joystick, and operates wirelessly if you like performing untethered. The chord voicing is a little basic, but with some judicious DAW and instrument programming it has the potential to generate engaging musical output or act as a handy additional MIDI controller.

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PolyChrome DSP McRocklin Suite

Guitar Rig Modelling Plug-in

Thomas McRocklin's first foray into the world of guitar rig emulations treads a slightly different path from the competition.

JOHN WALDEN

Thomas McRocklin might not be a familiar name for those outside the niche genre of instrumental guitar music. However, having done an opening slot in his hometown of Newcastle for Ozzy Osbourne at the tender age of eight, and then being taken under the wing of guitar virtuoso Steve Vai, his musical career started early in life. More recently, Thomas has created a YouTube presence and an online musical education programme, established relationships with guitar brands such as Kiesel and Fishman, and released original music as part of McRocklin & Hutch. His latest venture is a software-based guitar rig modelling system that's developed under his own PolyChrome DSP brand.

As a virtual guitar rig, McRocklin Suite's role is a familiar one, but there are a number of unique aspects to the design. First, with a couple of notable exceptions, the range of virtual gear included here means it makes a good case for being your 'all-in-one' amp modeller — it caters for a very wide spectrum of potential tones, from acoustic all the way through to high-gain. Second, the UI is not obsessively skeuomorphic and the plug-in doesn't attempt obvious virtual recreations of specific amps, cabs or pedals. It does, however, remain very intuitive in use, even for more traditionally minded guitar players. Third, it offers ultra-low latency. Indeed, the documentation claims a latency of 0.045ms and, in my own testing, Cubase Pro 12's latency monitoring showed a latency of just 2 samples. That's impressive, making real-time tracking through this plug-in a very smooth experience, and suggesting some pretty efficient coding under the hood.

The GUI design may be more modern than most, but it's still intuitive and based around familiar guitar components.

Go With The Flow

While the GUI doesn't try to capture the appearance of an amp, cab or pedalboard, the layout of the modern design is instantly familiar. A control strip at the top caters for the input and output levels, access to the extensive preset list, a very usable tuner, a noise gate and the somewhat addictive Wide option. As with any virtual guitar rig software, you need to pay attention to the quality and level of your DI signal, although noise is only really noticeable with higher gain settings and the noise gate itself is very effective.

The rest of the upper half of the GUI provides buttons to toggle between the four different amp models, controls associated with each of those amps, and five buttons that toggle the bottom half of the GUI between pages for the various pedal/effects options. The signal flow follows the layout of these buttons, with the amp models themselves situated before the speaker chain section. The labels for the four amps, Acoustic, Clean, Edge and Gain, give an accurate sense of their respective flavours but no real indication as to whether each is based on a specific real-world amp.

In terms of the effects, the first page provides a very interesting compressor (the Heatpressor, with both tone and saturation options included) and a synth-based octaver that can be blended in with your main guitar signal. The latter might not be as flexible as the synth found in the Neural DSP Archetype: Rabea plug-in I reviewed last month, but it does sound great and is very easy to use. The next page offers four different styles of drive pedal, Attacker, Shredder, Riffer and Viber, that span light boost-style overdrive to full-on distortion. These can, of course, be stacked, just as you might on a real pedalboard.

The Speaker Chains page offers an interesting approach. For the three electric guitar amps, you essentially get a choice of seven pre-configured 'mix ready' options, rather than detailed controls to select between different speaker/cab types, virtual mics and placements, or load third-party IRs. This obviously deliberate design choice provides a very streamline user experience, and might be one means by which the low latency is achieved, but it's not at the cost »

The expensive mic everyone can afford

The Aston Spirit's open, natural sound and sparkling high-end detail have made it a firm favourite with countless A-list artists and professional studios, often in preference to far costlier 'classic' studio mics. As it happens, this particular world-class LDC doesn't actually cost the earth, so it's a no-brainer for the home studio too.

It excels on vocals and acoustic instruments, while the versatility of switchable polar patterns and a 10dB/20dB pad makes it equally suited to a wide range of other applications. And it looks a million dollars too. More at **astonmics.com**

ASTON MICROPHONES®

» of the actual sound — there are plenty of tonal options here. You also get Resonance, Air and both Lo Cut and Hi Cut controls to further shape the sound and, via the next EQs page, a choice between a four-band parametric or eight-band graphic EQ.

Finally, there's a Post-FX page, with reverb, delay, a simple chorus and a final compression stage. These all work well but the reverb and delay are particularly impressive, spanning both conventional treatments and a whole smorgasbord of more creative effects. If ambient guitarscapes are your thing, there are plenty of options to indulge that.

The Sound Of (Mc)Rock

There are perhaps a few effects that are absent and would be nice to have here. For example, there's currently no fuzz effect or wah-wah, and the chorus is the only modulation option on offer. But the list of virtual equipment that is available here has been very well chosen. Without it ever feeling like parameter paralysis might set in, it covers a huge amount of sonic ground, and proof of this comes from the extensive collection of supplied presets. These are neatly organised into themed categories such as Acoustic, Clean, Low Gain, Dry and Riff/Shred, and there's a cool selection of 'sounds like' presets whose titles provide obvious hints as to the guitar tone target.

The various presets based around the acoustic amp are impressive. Yes, used with a DI'ed electric, these are more akin to hearing a 'produced' acoustic sound, but they can be big, bold, and more than ready to find their way in a busy mix. The cleaner and low-gain electric options are also very good, and you can easily find something that delivers a more traditional vibe — even if you're not looking at a (virtual) Fender, Vox or Marshall-style graphic!

The Heatpressor and Synth Oct options are both excellent, and the presets are bountiful and organised helpfully.

The Speaker Chains page provides the user with a streamlined approach for selecting the style of cabinet and microphone modelling.

Whether you want conventional or experimental effects, the McRocklin Suite's ambience options provide plenty of flexibility.

Thomas' own guitar style definitely pays homage to his original inspirations such as Steve Vai, but it also brings things very much up to the modern day (think Tosin Abasi, Tim Henson, Mateus Asato and similar), and many of the presets reflect these references. So, for example, there are lots of very clean lead tones with prominent attack, and options for rich ambient textures, and there are also plenty of examples of saturated lead tones that are ideal for speedy legato-style shredding. If these kinds of guitar tones are what you're chasing, the McRocklin Suite will be right on the money.

In use, while the amp models provide plenty of gain options on their own, the various drive pedals mean you can create almost any level of gain that you might desire. The plug-in does seem to respond well to playing dynamics, so you can control the overdrive with a combination of guitar volume and picking style. Of the other pedals, I found the Heatpressor particularly impressive, especially with cleaner sounds, while the reverb and delay are both excellent.

Also worth mentioning is that, whether used as a plug-in or standalone (where you get options for configuring audio and MIDI, as well as tempo), the low-latency operation worked just as advertised: I really didn't notice it.

Suite Choice?

If you're prepared to look beyond the established names, McRocklin Suite is most certainly worth exploring. Some users might initially find the lack of 'identifiable' amps, cabs or pedals a bit unsettling, while others might wish for a little more control over the speaker simulation. But these are design choices that have been made carefully and deliberately — in practice they do not limit the tonal possibilities at all. The overall result is a compact virtual rig that's easy on the eye, has an intuitive GUI, and offers both immensely flexible tone creation/sound design possibilities and low latency for real-time tracking. Sonically, it's fair to say that it's something of a Swiss Army knife, and it will most certainly appeal to more modern, technical guitar players, with its very impressive clean and high-gain tones. It's also competitively priced. Oh, and did I mention that it sounds very good indeed?

Sensibly for a new brand, PolyChrome DSP are happy to let you try before you buy; there is a free demo version available for download from their website. That's well worth doing. For those happy to embrace both the classic and the cutting-edge, the McRocklin Suite really is a bit of a gem.

summary

PolyChrome DSP's McRocklin Suite is an impressive virtual guitar rig with a slick, modern design and some excellent tonal possibilities.

$ $149.
W www.polychromedsp.com

When London’s legendary RAK Studios responded to demand and went immersive, they wanted the most truthful monitors available. So they chose The Ones Smart Active Monitors, finding them unbeatable for neutrality, imaging, fatigue-free listening and consistent mix translation.

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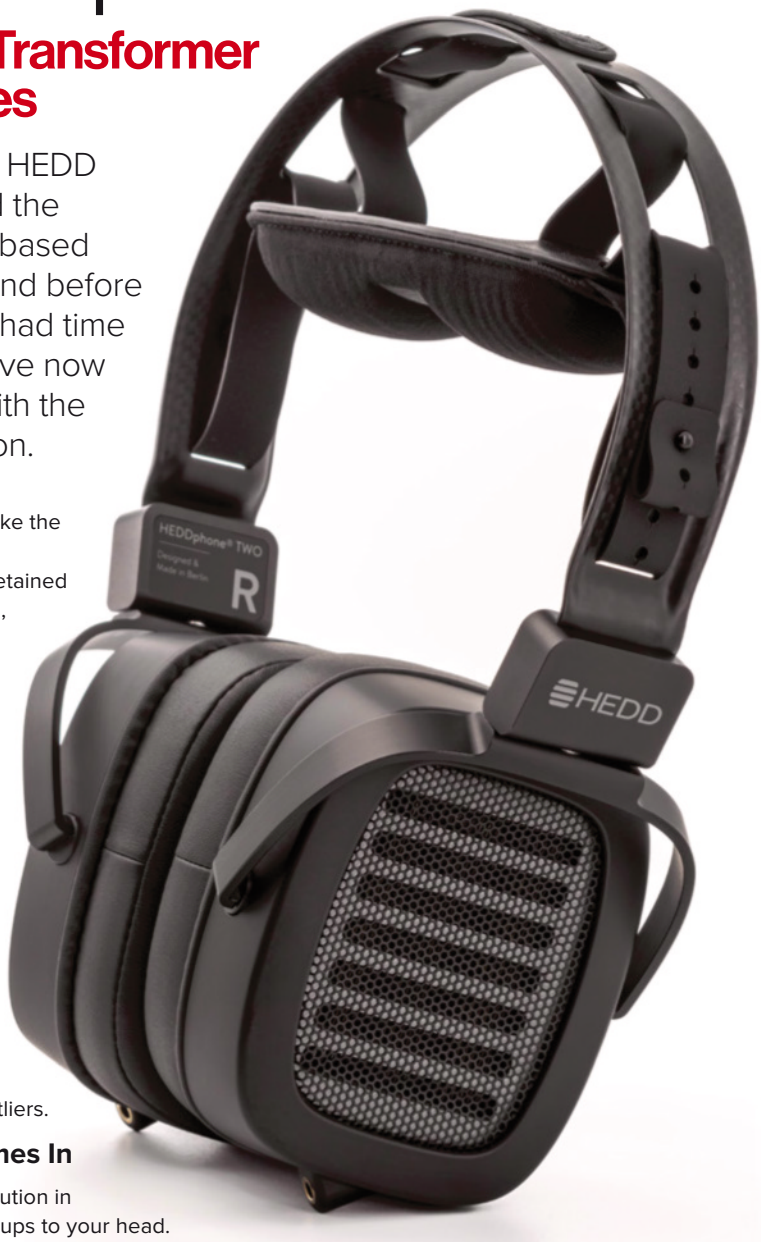
HEDD Audio HEDDphone TWO

Air Motion Transformer Headphones

SAM INGLIS

Three years ago, HEDD Audio developed the world's first AMT-based headphones — and before anyone else has had time to catch up, they've now replaced them with the second generation.



It can be disconcerting when your favourite products are superseded by new versions. When even the manufacturer acknowledges there's room for improvement, maybe the original wasn't quite as great as you thought! Or, worse still, the Mk2 could turn out to be one of those paper upgrades that never quite sounds as good as the old one. Either way, you're going to be left with a piece of kit that's worth less than it was yesterday, and an awkward decision about whether to upgrade.

Among my own longstanding favourites are HEDD Audio's unique HEDDphones. Sometimes called a 'ribbon tweeter', the Air Motion Transformer is used as a tweeter in several loudspeaker ranges, but HEDD are the only manufacturers who have managed to design a pair of headphones around it. The open-backed HEDDphones are bulky, costly, heavy, and inefficient — but they sound absolutely glorious. To my ears, they combine the sonic strengths of planar magnetic and moving-coil headphones, with deep and articulate bass, a very natural midrange, and a clear and open treble. Nevertheless, the HEDDphones are no more, and in their place we have the HEDDphone TWOs.

Shrinkflation

Since the launch of the HEDDphones, their designers have been beavering away to try to encapsulate the same sonic virtues in a smaller, lighter and more manageable package. With their huge oblong earcups, the original HEDDphones are the largest cans I've ever used, and whether or not you find them uncomfortable as a result — I don't, at all — there's no denying that they can make the wearer look a bit daft.

The earcup shape is retained in the HEDDphone TWOs, but the dimensions are noticeably smaller. They're still big headphones that stick out several inches, but it feels less like strapping a pair of NS10s to your head. And with the reduction in size comes a reduction in mass: at 550g, the new version is around 25 percent lighter. If you've tried typical models from manufacturers like Audeze and Hifiman, these won't seem like outliers.

'Til The Band Comes In

There's also been a revolution in how you strap these earcups to your head. The first-generation HEDDphones used a pretty conventional headband, with the only adjustment on offer being the ability to make this longer or shorter as required. The HEDDphone TWO employs an entirely new, patent-pending design called the HEDDband, which has multiple modes of adjustment, allowing not only the height but also the width and the pressure of the headband to be set by the user.

This is easier to understand by watching the video on HEDD Audio's website than from a written description, but in essence, the outer, structural headband contains within it a floating, padded 'roof' that sits on top of your head. Straps emerge from either end and from the top of this 'roof' piece, pass through holes in the outer frame and are held in place by small pegs on the outside. By making these straps longer and shorter, it's possible to control how far the 'roof' sits below the structural

headband, and also how forcefully the earcups are pulled inwards.

It's an ingenious system, and effective too, but you need to take the HEDDphones off in order to adjust the straps, so it can take a few attempts to find the optimal setting for any given head. Since the straps are secured using holes spaced nearly a centimetre apart, it seems at first as though the range of adjustment will be too coarse, but in practice, that's not the case at all, and even if you adjust both sides at the same time, a difference of one hole either side is quite subtle. I had no difficulty achieving a setting that felt comfortable to me — but, having been perfectly happy with the comfort of the original HEDDphones, I didn't notice a radical improvement in the TWOs.

Something that is a major improvement, however, is the provision of a high-quality, lightweight, semi-rigid case with a carrying handle. When you're spending this much money on a pair of headphones, you want to be able to keep them safe when not in use! HEDD Audio are clearly confident that they are built to last, too, because they're offering a five-year guarantee.

Like the originals, the HEDDphone TWOs use three-way connectors at the earcups to cater for balanced signals from headphone amps that can supply them. This time around, however, the cables attach using TRS mini-jacks rather than mini-XLRs. Two cables are supplied: a conventional unbalanced one terminating in a quarter-inch jack, and a balanced one with a four-pin XLR at the other end. These have their own little nest within the case.

The HEDDband permits an unusual degree of flexibility of adjustment.

Next Level

Having succeeded in making the HEDDphones smaller, lighter and more ergonomic, have HEDD Audio also managed to maintain their sound quality? The answer, in the best possible way, is no — because they haven't merely matched the sound quality of the original HEDDphones. They've bettered it.

With an impedance of 41Ω and a sensitivity of 89dB/W, the specifications are similar to those of the originals, but HEDD say they have significantly improved the performance of the Air Motion Transformer in the TWOs, and this is borne out in their subjective quality. It's not a night and day difference, or one that's equally apparent on all material; they still sound like HEDDphones. But, side by side, I consistently preferred the new version — which is impressive, considering how much I like the originals!

The Air Motion Transformer generates sound from a diaphragm which is folded like the bellows of a concertina.

If I had to sum up what I love about the HEDDphones it's that, paradoxically, they are the least headphone-like headphones I know of. Working with them feels more like working on a high-quality speaker-based monitoring system than is the case with almost any other cans. The HEDDphone TWOs simply take this even further, as though you're working on an even better pair of monitor speakers in an even flatter room. The originals still present a marvellously clear and balanced sound, but the TWOs are somehow even smoother through the sibilance range. At the bass end, meanwhile, low-frequency extension is impressive on both, but there's an additional richness to the TWOs that somehow connects the bass to the low midrange more naturally, without ever sounding overblown or exaggerated. And, of course, all the existing strengths of the HEDDphones are retained.

When I reviewed the originals, back in 2020 (www.soundonsound.com/reviews/hedd-audio-heddphone), I said that hearing music on them is a bit like being in a good mastering environment. They're extremely revealing, but not in a brutal way. If you've overcooked the low end, overlooked sibilance or pushed the midrange too hard, they'll administer the most polite of taps on the shoulder to let you know! And if the mixes are good, listening to music on them is pure heaven. Many times during the review period I put on test material to audition some specific aspect of their performance, and half an hour later found myself buried in a favourite album.

In their new incarnation, the HEDDphone TWOs are still relatively bulky, still quite heavy, and still require a good headphone amp. And, sadly, they still aren't cheap. But they remain very much one of my favourite products — and if you're considering a pair of high-end headphones for producting, mixing or just getting lost in music, they need to be on your shortlist.

summary

HEDD Audio's innovative AMT-based headphones have got smaller, lighter, more portable and more adjustable — and they sound even better.

$ €1999
W http://hedd.audio

Zoom MicTraks

Handheld 32-bit Recorders

Once again, Zoom raise the bar for high-quality, affordable handheld recorders.

HUGH ROBJOHNS

In 2019, Zoom brought us the world's first location recorder that offered 32-bit floating-point recording, the F-6. This revolutionary technology removes the need to optimise recording levels in real time, so is of particular benefit to inexperienced users or those working in situations where levels can't be controlled or predicted. Other Zoom models (and other manufacturers) soon adopted the same technology, and the latest to do so are Zoom's new MicTraks, reviewed here.

A range of low-cost, portable audio recorders, the MicTraks — the M2 stereo recorder, the M3 stereo on-camera recorder, and the M4 four-channel recorder — share common features, including the F-series' dual-converter 32-bit floating-point recording technology. Powered by AA batteries (two in the M2/M3, four in the M4) or over USB, they record directly to micro-SD cards (up to 1TB SDXC) and can serve as USB mics when connected to Windows, Mac, iOS, or Android devices. While the M3 mounts directly onto a video camera's hot shoe (see 'Rule Of Three' box), the M2 and M4 resemble large stage mics and are clearly intended for handheld use.

Two Cool

The M2 MicTrak is a stereo recorder, with integral microphones protected under a large open-mesh plastic grille. It's 221.5mm long, with a 62mm diameter at the grille reducing to 30mm at the base, and with batteries installed weighs just 204g. Included accessories are a furry windshield, a large plastic mic-stand clip (with 3/8-inch thread adaptor) and several Quick Tour guides in different languages. The guide is sufficient to get up and running, but a detailed online manual provides useful background and details of many facilities not mentioned in the guide.

Arranged in a coincident 90-degree X-Y format, the mic capsules were designed specifically for the M series and give a wide stereo recording angle (SRA) of around 180 degrees. A summed mono recording mode is also available. One advantage of 32-bit floating-point recording is that there's no need for a manual gain control — the M2 can cope with anything up to a whopping 135dB SPL — so there isn't one.

The M2's body is made of a shiny, smooth, black plastic, designed to minimise handling noise, and the top surface is dominated by a small monochrome 96x64 LCD screen. Buttons below select mono/stereo recording, activate a high-pass filter (off, 80, 160 or 240 Hz), and scale the waveform when the Home screen is showing. When the Menu screen is activated by a button on the right side panel, the buttons navigate the menus and adjust parameters.

The transport controls on top form a quincunx with the large red record button at the centre, and fall easily to hand. Operation is logical and familiar, and all the expected functions, including a 2s pre-record buffer, can be found easily in the simple menu.

A knurled thumb-screw releases the battery compartment, which slides from the base of the recorder to accept alkaline, NiMH or Lithium batteries. Alkaline batteries give around 11 hours continuous operation, while Lithium cells almost double that, and the unit powers off automatically after a selectable period of inactivity (sensibly, it doesn't switch off if recording or connected via USB).

The M2 can be employed as a USB mic with a fixed 48kHz sample rate and 24- or 32-bit audio data and, usefully, the M2 can record to its micro-SD card while live as a USB mic. An ASIO driver (from Zoom's website) is required for Windows, but not for Macs, smartphones or tablets, and a card-reader mode allows a computer to access any recorded files over USB, which is probably easier and safer than extracting the card!

Recordings are always 32-bit floating-point WAVs, at sample rates of 44.1, 48, or 96 kHz. Long recordings are automatically split at the 2GB limit and saved to the micro-SD card at regular intervals, so a power failure won't lose an entire recording. Files are identified by a date stamp and take number (incremented for each new recording made on that day). If necessary, very quiet or loud recordings can be normalised through a menu function and saved as a copy file for export in 32-bit floating-point, or 24- or 16-bit fixed-point formats. If recorded files need to be synchronised to video, there's a menu option to record a half-second tone blip at the start of each new take: connect the M2's headphone output to the video camera's input, and that blip will make sync'ing much easier in post-production.

The monitoring system's auto-volume

Zoom MicTraks

PROS

- 32-bit floating-point technology: no need to set recording levels!
- Impressive sound quality.
- High-quality preamps.
- Easy to use.
- M4 adds useful I/O and timecode facilities.
- Good battery life.

CONS

- USB cable not included.
- Prone to excessive LF noise without HPF and windshield.

SUMMARY

These high-quality MicTraks raise the game significantly for budget audio recorders.

feature makes life easy with loud and quiet recordings. Monitoring during recording and playback is via a 3.5mm stereo headphone socket on the left-hand side with up/down buttons to adjust the volume. Just 20mW of power is available per channel, but I found that adequate with efficient 32Ω headphones. There's also a tiny built-in speaker for playback auditioning.

I found the waveform display reassuring and helpful, and with 99 location marks available navigating files is simple. Although tiny, the screen shows all the relevant information on timings, battery status and so forth during recording and playback.

Using the M2 is pleasingly straightforward and in general recordings were quiet and to a high standard. If holding the recorder, the low-cut filter and windshield are essential to prevent unwanted LF wind and handling noise, and I heard an occasional HF noise when I placed the unit very close to a Wi-Fi modem or phone, but the M2 is a great machine for recording podcast interviews, live fly-on-the-wall musical performances, or natural ambient sound effects.

Rule of Three

The M3, not sent for review, is intended exclusively for video camera applications. Based around the same internal tech as the other MicTraks, it has a Mid-Sides stereo mic with a short interference tube in front of the Mid element, and a switchable low-cut filter mode is available. The mic comes with a hot-shoe shockmount, a foam windshield, and a mini-jack cable to link decoded stereo sound directly into the camera as a backup and for sync. We're told a pair of AA batteries has a 12-hour continuous working life, and a headphone output allows direct live monitoring and checking of recorded files on playback. Four audio tracks are recorded to micro-SD: the raw M-S outputs plus decoded L-R stereo. The equivalent stereo recording angle (SRA) of the latter is configurable between mono, 90 and 120 degrees. The only supported sample rate is 48kHz (fine for video) and the only file format is 32-bit float (WAV/BWF), but a free app allows fully adjustable M-S source decoding in post, and offers level normalisation of the 32-bit files.

The M3, mounted on a camera.

Fantastic Four?

The M4 MicTrak is essentially a four-track version of the M2, measuring 206mm in length, 70mm at its widest part, weighing 325g (with batteries), and costing twice as much. The shiny, low-noise plastic body, controls, menus, USB connectivity, stereo mic array, and specifications are all virtually identical. However, the M4 offers extra physical connections and the buttons to activate them, built-in timecode, a larger (240x240) colour LCD screen, the ability to record at 192kHz (as well as 47.952 and 48.048 kHz for video work), and a 6s pre-record buffer. Included with the M4 are a pair of fabric (not furry) windshield covers and, again, Quick Tour guides in various languages, but there's no stand-mount clip: this device has a 1/4-inch threaded socket on the bottom to mount on a camera tripod.

On each side of the unit are combi XLR sockets for the additional two inputs. These accept up to +4dBu on the XLR (mic) input or +24dBu on the TRS (line) input, and phantom power is available at 24 or 48 V. The preamp circuitry is apparently derived from Zoom's F-series products and performs very well (-127dBu EIN with a 150Ω source). If that weren't enough, the M4 also has four further connectors. Two 3.5mm sockets accept a stereo mic/line input (with plug-in power) and provide a stereo line output (0dBFS=+1dBu), while a third is used for timecode in/out, with the usual sync options and supporting all standard frame rates. Finally, a 2.5mm socket accepts remote control from Zoom's RC2, RC4, RCH-5, or RCH-6.

Naturally, with four tracks the display needs to be larger and convey more information, but the colour helps and it's easily readable under normal lighting. Again, the sound quality is impressive, both from the internal mics and the external inputs, and the compact size and timecode options make it ideal for video work. The ability to record additional balanced mic or line inputs (eg. from a FOH mixer) adds considerable versatility. When recording mono or stereo tracks from the M4's internal X-Y mic array, the AA batteries provide around 19 hours (alkaline) or 31 hours (lithium) of life, and using all four channels reduces this to a still-impressive 10 and 18 hours, respectively.

Right On Trak?

I'm very impressed with the MicTrak recorders, which offer remarkable value for money and genuinely good sound quality. It would have been nice if USB-C cables were included, but that's a very minor objection and at this price there's really nothing else to complain about. The M4 looks professional and reliable and can do a lot. Connecting XLRs to the sides looks ungainly, and right-angled plugs might be a good idea to minimise leverage on the sockets, but the resulting recordings are impressive. The smaller M2 may lack some of its sibling's features, but it's a very capable recorder indeed. Both models deserve consideration by anyone looking for high-quality budget recorder.

$ M2 $199.99. M3 $199.99. M4 $399.99.
W https://zoomcorp.com

Shure SM7dB

Active Dynamic Microphone

Shure's classic mic celebrates a milestone birthday with the addition of a built-in preamp. We test the new SM7dB and explore its fascinating history.

SAM INGLIS

The Shure SM7B has never been reviewed in *Sound On Sound*, because it's older than the magazine itself. It reaches its half century this year, and thanks to its popularity in podcasting and live streaming, does so as one of the few professional microphones that is instantly identifiable to everyone. To mark the occasion, Shure have introduced an active version.

Moving-coil dynamic microphones are typically much less sensitive than capacitor mics, but at just 1.1mV/Pa, the SM7B is a gentle giant even by dynamic mic standards. This is partly because it was designed before powerful rare-earth magnets were available, and partly because Shure's engineers set out to maximise other aspects of its performance such as frequency response instead. This led them to omit the transformer found in other models such as the SM57, which provides 'free' gain but curtails the bass response, and to position the diaphragm some way behind the grille.

These decisions certainly paid off in terms of sound quality. In a studio context, the SM7B earned a place among the relatively small number of moving-coil dynamic mics that were perceived as able to hold their own against capacitor models. Its lack of sensitivity was seldom a problem in a professional environment, and was even an advantage in roles such as close-miking bass amps and kick drums.

For its first 35 years or so, the SM7B sold pretty much exclusively into studio and broadcast markets, earning its keep but probably not making much impact on the bottom line. Then came podcasting, live streaming, YouTubing and other neo-verbs of the late 2000s. Suddenly, the Shure SM7B was the must-have mic for anyone talking to other people on the Internet.

However, when people with cheap mixers and audio interfaces started trying to use their new mics to talk to other people on the Internet, they sometimes ran into a problem. Even with the input gain maxed out, they were struggling to get enough level out of their SM7B — and those cheap mixers and audio interfaces rarely sounded great with the input gain maxed out. Enter the enterprising entrepreneurs at Cloud, and the rest is history. The idea of a simple, fixed-gain, inline preamp for moving-coil and ribbon mics proved to be genius, to the point where it's now almost expected that people buy a Cloudlifter along with their new SM7B.

Cloud Cover

In essence, what Shure have done with the SM7dB is to formalise that connection. Their engineers have worked with Cloud to design a phantom-powered preamp circuit that could be integrated into the mic itself, to all intents and purposes providing a built-in Cloudlifter. This is not quite as trivial as it sounds, since the internal space within the mic serves an important acoustic function, and filling it too full of circuitry would risk changing the sound.

The result is a mic that is outwardly pretty much identical to the SM7B, until you look at the end plate. The high-pass and mid-boost filters from that mic are present and correct, but they're no longer toggled using recessed switches that need a sharp implement to move them. The new switches sit proud of the surface so that they can be moved with a thumb, and there are now four rather than two. One of the new switches engages the preamp, whilst the other toggles between two fixed gain settings: +18 and +28 dB. Given that the SM7 isn't a mic you'd often use handheld, I don't think there's too much risk of accidentally nudging these switches, though the new arrangement doesn't make the filter settings quite as visually obvious as the old one.

Shure SM7dB

$499

PROS

- Delivers a simple, foolproof way of raising the sensitivity of the classic SM7B.
- The active circuitry is sonically transparent and can be bypassed.

CONS

- None.

SUMMARY

If you love the sound of the SM7B but wish its output was a little hotter, the SM7dB is for you!

The end panel of the SM7dB features four switches: two to engage the built-in filters and two relating to the new preamp circuit.

The SM7dB ships inside an outsized cardboard box. There's no further case or bag supplied, but you do get a spare removable windshield. It attaches to the mic stand in exactly the same way as the SM7B, using an integrated yoke which also houses the XLR output connector.

Shure say that, "The integrated preamp in the SM7dB was designed ground-up by Shure to couple directly with the SM7B cartridge and accurately/transparently amplify it — delivering consistent gain, flat frequency response and extremely low noise." When it's switched out of circuit, the dB should be effectively identical to the SM7B. When it's engaged, it should provide a transparent gain boost that doesn't otherwise change the sound. And that's exactly what it does.

With the preamp switched out, I could detect no difference between the SM7dB and my own SM7B, either in terms of sensitivity or tone. Engaging phantom power and switching in the preamp simply made the SM7dB louder. Assuming you are using a colourless, neutral preamp such as are found in most audio interfaces, the results are indistinguishable from simply turning the preamp gain up. In effect, it's like a pad in reverse, extending the gain range of your mic preamp by 28dB at the upper end — which, in the case of the SM7, is where it's going to be useful. With coloured or character preamps, it also gives you the option to hit the front end of the preamp much harder than would be possible with the passive SM7B. (It should be pointed out, though, that if you have good clean preamps with plenty of gain, there's no advantage in terms of noise to using the dB's active circuitry or indeed any inline preamp.)

The SM7dB is a totally logical evolution of the SM7B. The built-in preamp will be genuinely useful in many real-world recording, streaming and broadcast uses — and if you don't need it, you can simply switch it off. Or, of course, you can just buy an SM7B, which remains a current product.

The SM7 Story

There's really not much more to say when it comes to reviewing the new version, but like many people, I'm curious about some of the myths that have grown up around the SM7 and its history. Shure very kindly made their historian, Michael Pettersen, available for interview, and I began by asking him about the mic's origins. The moving-coil element within the SM7 is directly descended from the first ever single-element cardioid dynamic mic. "Inside of the SM7 is a Unidyne cartridge — a Unidyne 3, the third generation — but it goes back to Ben Bauer and the introduction of the first Unidyne in 1939. You could do directional microphones »

» before that, but it required two elements and that was expensive and hard to tune. Bauer's breakthrough at age 23 was to figure out how to use acoustically delayed signals to create that directionality. The idea is to take the wave that's coming from behind, acoustically delay it ever so slightly and then bring it to the back of the diaphragm. It's still employed in virtually every directional mic today. It was kind of an axiomatic concept that people still use. Shure and everyone else!"

Ben Bauer left Shure in 1957, and the next step in the evolution of the Unidyne capsule was made by another young engineering genius named Ernie Seeler. The Unidyne 1 and 2 cartridges had been suited only to side-address mics, but in 1959, Seeler's Unidyne 3 changed that. Introduced that year, the Shure 545 is still a current product today — and the source of lots of online misinformation. "I get so amused by reading all the experts on the Internet who say that the 545s are rejected SM57s! In today's age, people will believe an influencer that has absolutely no knowledge whatsoever, but they won't believe the manufacturer. If it says Unidyne 3 on it, it's a Unidyne 3. The engine is the same. It's got a switch on it, and the output transformer's dual-impedance instead of single-impedance, but other than that — it's funny, every now and then I'll read something online: 'Just bought a 545, Wow, it sounds as good as an SM57.' Yes, it would..."

Five To Seven

Strange as it may seem now, the ubiquitous SM58 and SM57 were developed primarily for studio applications — indeed, the 'SM' actually stands for 'studio microphone' — and the SM7 evolved out of another model in this range. "We were trying to get our microphones into studios in the '60s, and one of the things we came up with was the SM5B.

"That used a Unidyne 3, and we put a really good shockmount on and we put really big windscreens on it, because we were trying to sell it as a boom mic for movies and television work. The reason we didn't have a condenser microphone is because we didn't have a condenser microphone. We didn't have any condenser technology. So we took the best that we had and hoped that it would work well for booming.

"Dynamic microphones just really aren't sensitive enough to do boom work, so the SM5 was a failure as a boom mic — but

Michael Pettersen is Shure's company historian.

somebody in radio stations started using it a close-up mic for on-air work and it had a great sound. And so we started to sell these things for radio stations and for television voiceovers as well. But the problem with radio stations and TV stations is they take care of their equipment. Once they've bought one or two mics, they don't buy any more because they've already got them, and so it was self-limiting. The SM5B is still really revered by many people in the broadcast area, but we never sold very many of them, because of the fact that it was too big to use on stage. It wasn't a live mic. It didn't really have the sound that you would use for miking the Rolling Stones in the studios. It was pretty much limited to voiceover, but there aren't that many radio stations or TV stations.

"We wanted to come up with something that perhaps had a bigger sales potential and might be used live or in recording studios. And so we kind of took the best aspects of the SM5 and tried to shrink it down and make it smaller. The idea was to make a less expensive, smaller SM5B with some adjustable frequency response on it. And that was the SM7, which was first announced in 1972, but not sold until 1973.

"And I'm certain Ernie Seeler was part of that, because it was a Unidyne 3 element, but one of the primary persons responsible for it was a guy named Gerry Place. Interestingly enough, he was a microphone development engineer, but he designed the way the SM7 looks. It was the only industrial design he ever did, and he got a patent on it. And it still looks modern even though it's 2023. He just managed to kind of catch this timeless look.

"So it comes out in '73, and it was... OK. Sales were so-so. Some were bought for studio voiceover booths, some were bought for radio stations. In the 1980s, Michael Jackson, it was used for his *Thriller* album. That was a big deal as far as PR goes, but it didn't really affect the sales. And really, for about 35 years, it just was kind of an average seller that was never a hit."

Fine Margins

Much Internet mythology has grown up around the design of the SM7, and especially the differences between that mic and the more humble SM57. "We get people asking how to turn an SM57 into an SM7," laughs Michael. "And this guy says 'You take out the transformer in the back of the SM57 and remove all the potting material, and all of a sudden your SM57 is an SM7!' No. You ruined your SM57, but it's not an SM7 now."

Although the cartridge in both mics is designated Unidyne 3, that doesn't mean it's exactly the same. Subtle changes needed to be made to adapt the capsule to work without the SM57's transformer. "The transformer not only changes the impedance but also kicks up the level. So, the SM7 coil has more windings on it. This makes it a little bit harder to move, if you will, because it is heavier. But it's also the reason that you've got to kind of be on top of the microphone, and sounds from farther away don't really pick up that well, because it's just harder to get that diaphragm and voice coil to start moving because of the extra mass."

Does that not also change the high-frequency response? "A little bit," agrees Pettersen, "but we can offset that with resonators and so forth. Also, the flexibility of the diaphragm is a little bit different. It's not the same Mylar. It has a different flexibility, which affects the low end, and of course, you've got the bigger acoustical space behind it. So you combine all those things and it affects the low-frequency response. You know, microphones move millionths of an inch, so it doesn't take much of a tweak anywhere to really change how it sounds.

"When you get down to what the differences are, it's just miniscule stuff. And people say 'The SM7 doesn't have enough output!' the reason is the diaphragm is set back from the edge of the microphone. Primarily we did that to control proximity

effect and to control 'p' popping. If it was forward, right up the edge, where the grille is, I don't think people would like it as much — but because of that, you gotta have a little more gain than folks are used to."

From A To B

Another topic that has provoked online discussion is the differences between SM7 variants. As Michael explains, these reflect functional development that made little discernible change to the sound of the mic.

"The SM7 came out for sale in 1973, then the SM7A came out in 1999. And the only difference was we added a much more effective humbucking coil, because radio stations were replacing paper copy with computer monitors and they were placing them very close to the microphones. And certain computer monitors did not have very good shielding and the SM7 would pick up hum. So we had to come up with a better anti-hum mechanism, to put in the SM7. So that was the A.

"The B came about only three years later, and we did that primarily for two reasons. They were being used now by people who didn't really know mic technique that well, and so better pop protection would help, but the other reason was to make it look like the SM5B, because there were radio stations and particularly older engineers that just said, 'You guys are so stupid. That was the best microphone we ever had. Why did you discontinue it? And one of us said, 'Maybe if we make it look more like the SM5B, It would sell better,' and sure enough, it did. So we put the fat windscreen on there, and it helped you work closer, but also made it look more like an SM5B."

Lucky Seven

The SM7 found a home in many recording and broadcast studios, and remained a solid but unspectacular seller for 35 years or so. And then... "In 2007-2008, all of a sudden, SM7 sales started to climb. We do a little research and we find out, this thing called podcasting is taking off. We were not advertising it for podcasting. I doubt we probably knew what podcasting was! And then, of course, because podcasts are done at home studios, there's far more of those than there are radio studios — and there's also a lot of podcaster see, podcastser do."

Perhaps the oddest thing about this whole episode is that it's not the first time something like this has happened. "The SM58 was not a success for about 10 years. We were trying to push this as a radio and TV interview mic, and all of a sudden in the mid-'70s, it starts to take off and we said 'Man, people are using this for live sound. What's going on?' Many times, our history is that we're in the right place at the right time with the wrong tool, or we're talking to the wrong people. And the market comes around, says 'Hey! This is really what you want it for!' And that's what happened with podcasting.

Michael Pettersen: "In 2007-2008, all of a sudden, SM7 sales started to climb. We do a little research and we find out, this thing called podcasting is taking off. We were not advertising it for podcasting. I doubt we probably knew what podcasting was!"

"So, you know, the podcasting did not surprise us. We were really shocked when gamers started using it, though. Gamers are gonna pay that much money for a mic? Still, you can't deny it sounds great, and certainly when you're in a room that's less than perfect, it does tend to pick up less background noise than a condenser microphone does. And so you combine all those things, and it's just been great. I mean, we just opened up another manufacturing facility, and a lot of it is dedicated to making SM7s. In 2023, compared to 20 years ago, we sell probably about a hundred times more than we used to!"

Though its makers were unaware, the SM7B was clearly the right mic in the right place at the right time. It's not hard to imagine that it could still be available another 50 years from now — and who knows what unexpected uses people will find for the new active variant?

$ $499
W www.shure.com

Rupert Neve Designs MBT

Stereo Bus Processor

BOB THOMAS

Rupert Neve Designs (RND) are probably best known for their mixers, mic preamps, EQs and dynamics processors, but they've also built up a significant range of master-bus related products, and their latest release is the MBT, or the Master Bus Transformer to give it its fully expanded title. The MBT is a stereo device that's designed to give the user control over a range of potentially pleasing processes, including tone-shaping, compression, stereo width enhancement and harmonic distortion.

For the first time, the Red and Blue bands of RND's popular Silk circuit can be used together in the same device — but there's more to the MBT than purple Silk!

Construction & Controls

The fascia of its substantial, 2U 19-inch rackmountable, black-painted steel chassis offers plenty of space between the controls, and the white legending is large enough to be easily readable from a distance (something that can often present a challenge to spectacle wearers such as myself!). On the back, you'll find XLR connectors for the electronically balanced inputs and transformer balanced outputs, along with a combination IEC connector and mains power on/off switch. And on the inside, as with all the RND products I've reviewed, the PCB design, layout, components and build are all of the highest quality. The main PCB is dominated by the two channels' custom output and inter-stage transformers, and power to both it and the front-panel daughterboard is provided by a pair of high-quality switch-mode power supplies.

The front-panel control sections feature 31-detent potentiometers throughout. Those with dark blue knobs set operational parameters, while those with red knobs (with one exception) control gain. All function in/out switching is carried out using relays activated by buttons that (with a few exceptions) illuminate green when active. A non-illuminated button selects between the low and high compression ratios.

The input section features a ±12dB trim control and a relay-switched high-pass filter, whose corner frequency can be set within a range of 15-100 Hz, allowing you to remove unwanted low or subsonic frequencies below the cutoff point. The post-trim/pre-EQ stereo signal level is displayed on a pair of LEDs that turn green at -20dBu and red at +23dBu.

The relay-switched EQ that follows the input is a simple two-band shelving setup with up to ±9dB gain per band. The low-frequency band covers 30-240 Hz and its high-frequency counterpart 3-24 kHz but, as you'd expect, these curves reach deeper into the frequency spectrum than those figures might suggest, and the bands overlap somewhat between approximately 350Hz and 2.5kHz.

On leaving the EQ section, the stereo signal enters the MBT's relay-switched

Color Comp. This programme-dependent optical compressor has a fixed 20ms attack time and simplified control set, with threshold (0-24 dBu), release (100ms to 1.5s), a simple button switch to select a low (2:1) or high (5:1) ratio, and up to 20dB of Class-A make-up gain. A high-pass side-chain filter with a variable corner frequency (20-350 Hz) allows you to prevent low frequencies from triggering unwanted compression. A green LED illuminates when gain reduction is taking place, and a dry/wet Blend control makes dialling in just the right amount of parallel compression a breeze. The coloration that gives this Color Comp section its name comes, according to RND, from the specific kinds of low-order distortion and other non-linearities that are generated when using an LDR (light-dependent resistor) for compression.

A potential side-effect of compressing a stereo source can be a narrowing of its soundfield, and the relay-switched Width section that follows allows you to compensate for this, but it also gives you the ability to creatively widen your mix's stereo image. It's not the simple M-S balance control you might have expected, though. It operates on a sum of the L-R signal, which is high-pass filtered and then passes through a gain stage. The signal is then multed, with a polarity inverted version fed to one of the L-R channels and a non-inverted one to the other. The Width knob controls the gain stage, while the always active high-pass filter has a user-adjustable corner frequency range of 50-800 Hz, so you can remove low and low-mid frequencies up to 800Hz from this 'synthetic Sides' signal to avoid any low-frequency anomalies that result from the process.

Super Silk

Super Silk is an updated implementation of RND's signature Silk Texture function that allows you to add a variable level of second- and third-order harmonic transformer-based distortion into the output signal. In other products, there are two colours, Red Silk and Blue Silk, and these are both present on the MBT. But, I believe for the first time in an RND product, they can be accessed individually or in combination, an arrangement that the indicator LED suggests we might call Purple Silk! Each Silk has its own individual level control knob (a very dark blue for Blue Silk and the standard red for Red Silk), allowing you to create the perfect harmonic distortion balance for your mix. The Super Silk section's Harmonics control determines the overall amount of low-order harmonic distortion.

Super Silk utilises a dedicated L-R pair of inter-stage transformers that can be pushed close to core saturation without affecting the output level. A Drive LED sitting between the Blue Silk and Red Silk level controls illuminates when the signal level entering the inter-stage transformers reaches the optimal range for the production of the Silk harmonic distortion. This new Super Silk circuit topology allows you to achieve much higher levels of harmonic distortion than were possible under previous iterations of the Silk Texture function. In addition, an amber-illuminated button activates »

Rupert Neve Designs
Master Bus Transformer
$3999

PROS
- Extremely impressive performance.
- Can generate high levels of harmonic distortion.
- Mastering quality optical compression and EQ.
- The stereo widening function works really well.

CONS
- Not inexpensive.

SUMMARY
A very impressive combination of equaliser, optical colour compressor, stereo widener and harmonic distortion generator that delivers an overall performance that justifies the professional price tag.

The electronically balanced inputs are on Neutrik Combo connectors, while the transformer-balanced outputs are on XLRs.

» Zener diode-based soft-clip circuitry that sits after the inter-stage transformers and adds emphasis to the Silks' harmonic distortion.

In the final front section there's ±12dB of output trim available. This area also hosts a red-backlit button that relay-switches the MBT into hard bypass, as well as a twin 16-segment LED output level meter, scaled from 0-24 dBu, and a green power-on LED. The evenly spaced meter scaling is worthy of mention: with 0-12 dBu being covered in two 6dB steps, 12-16 dBu in a single 4dB step and 16-24dBu in four 2dB steps, there's helpfully higher resolution across the most critical area.

On Test

Once you've inserted the MBT on your master bus, everything that follows requires only your ears, intuition and creativity. The high-pass filter on the input is an extremely effective subsonic filter and I found myself leaving it permanently in circuit, set at approximately 25Hz. The equaliser is smooth, transparent, effective and sounds extremely musical to my ears. The high-frequency band's extension up to 24kHz and beyond offers the ability to add an airy, silky high end to a track, whilst the low-frequency extension meant that with judicious use, and careful balancing with the high-pass filter, I could boost a track's low-end foundations and still maintain definition.

As a master bus compressor, the MBT's Color Comp does the business too. Set as I like my master bus compressor to be (a 2:1 ratio with the side-chain HPF engaged to keep unwanted low-frequency-driven compression in check, with a long release time and the threshold set so the LED flickers to indicate just 2-3 dB of gain reduction) it sounded good. Really, really good. With tracks that needed some compressor 'glue' to hold them together, increasing the amount of compression and, where necessary, increasing the ratio to 5:1, it gave me exactly what I wanted. But there's more to it than that. The higher compression level built up the track's weight and warmth with the increased low-frequency harmonic content. And, of course, the Blend control made setting up parallel compression simple and intuitive when required.

If I have a favourite section on the MBT, though, it's the Width processor. I'm something of a convert to M-S

The curves of the shelving filters can reach well up/down into the audible band, even at their most extreme settings.

mastering but on heavier mixes I found that I preferred the sound of the MBT's Width process. With the Width control set at approximately 60 percent of its travel and the HPF at about 100Hz, the stereo soundfield of one of my favourite tracks (Colin Blunstone's live performance of 'She's Not There' from the *Night of the Proms '93* CD) was lifted out of my monitors in stunning fashion. I did experiment with boosting the Sides signal in my DAW to see how close this would get me, but while it gave me a result that I wouldn't have been reluctant to go with it just didn't feel quite as exciting as what I'd achieved using the MBT's Width control.

The Silk harmonic distortion facility has always appealed to me, but the increased level range available within Super Silk's two frequency bands (in broad terms, across low to low-mid for Blue Silk and high-mid to high for Red Silk) mean you can now make things really saturated if you feel the need. As I've found in the past, deciding on the amounts of either Silk is a matter of using your ears rather than anything else. Since the MBT offers the ability to run both colours separately or in combination, and the Harmonics knob gives you control over just how much of that Silking drives the inter-stage transformers, you might find you want to spend a little longer listening out for exactly what this section is doing.

That said, you might want to keep an eye on what the Drive LED is doing, since that gives you a measure of what's happening inside the inter-stage transformers. A solid (and pretty bright) LED indicates that you're getting very close to core saturation, while for mixing and/or mastering if the Drive LED is flickering and the Red and Blue Silks and/or the Harmonics master are in the first 20 percent of their travel, you should be in a fairly safe zone, where transients aren't being reshaped too much.

Switching in the transient-shaping Zener diode soft-clipping acts to increase the overall harmonic saturation and bring out further the impact of the Silk mix that you have created. Personally, then, I think this all adds up to make the MBT as good a candidate for processing things like drum, synth and guitar subgroups as for master bus processing or mastering.

Finally, it isn't often that the output stage of a piece of outboard gets singled out for mention, other than in relation to any potential transformer-related distortion, but in the case of the MBT the

The optical Color Comp compressor features both a side-chain high-pass filter and a parallel compression Blend control.

The Super Silk section allows you to add more distortion than in previous products, and to blend the Red and Blue Silk characters.

±12dB output trim control is a critical component. Sitting in the signal path are five potential sources of gain — input trim, EQ boost, the Color Comp's make-up gain, and the additive contributions of the Width and Super Silk sections — so in practice the output trim sees pretty much constant use, as you balance the effects of the first four gain stages, the amount of Silk you add, and the input level requirements of the next unit in your signal chain.

Ride The Bus?

Rupert Neve Designs' Master Bus Transformer is an extremely impressive device that delivers exactly what its designers intended it should, and although the price tag puts it out of the reach of all but professional studios and wealthy project-studio owners, its audio quality and performance easily justify that cost. Obviously, signal coloration is a matter of taste, but if your production direction and technique involve adding copious amounts of harmonic distortion to your subgroups, stems or mixes, the MBT is definitely going to appeal. Special mention also goes to the Width function, which does broadly what M-S based stereo widening can do but in a slightly more exciting fashion, especially when processing heavier tracks. I'd urge anyone who has both the need for bus or mastering colour and the budget available to audition an MBT at the earliest opportunity. I'll be very sorry to see this one leave!

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Cherry Audio Software Synthesizers
Novachord & Solovox

Cherry Audio delve into the distant past to bring us two new virtual instruments.

GORDON REID

Although it appears simplistic to modern eyes, the Hammond Novachord, which introduced much that we now take for granted in analogue synthesis, must have been mind-boggling when it appeared in 1939. All of its sounds were generated from a single waveshape created by a fully polyphonic divide-down oscillator bank. The output from this was then passed in parallel through five filters, and the outputs from these were mixed with the unfiltered signal using the front-panel volume controls. Deep Tone was a path through a gentle low-pass filter, the three Resonators were paths through differently tuned band-pass filters, and Brilliant Tone was a path through a gentle high-pass filter. From here, the signal passed through an attenuator (the 'Balancer') that affected only the lower half of the keyboard to allow the player to balance the treble and bass registers to taste, while a further set of low-pass filters (controlled by the 'Bright-Mellow' lever) affected the upper 54 keys but left the 18 lowest notes unaffected.

Loudness shaping was performed by an ADSR contour generator. Seven preset shapes were provided, ranging from a piano-like AD contour, through a range of ADS contours, to vocal-like AS contours with slow attacks. In all cases, the release was instantaneous unless you pressed one of the sustain pedals. The final stage in the signal path was a simple volume control.

The modulation generator was perhaps the most unusual part of the design. This comprised no fewer than six mechanical, fixed-frequency LFOs, with each affecting two of the notes in the octave. Because these oscillated at slightly different rates and depths, they imparted a more complex and musical vibrato than would have been obtained by applying a single LFO to all 12 master oscillators. Four settings were available: off, small, normal, and small+normal for the deepest effect.

The only other control was a switch called Combination, which was, in effect, a two-position patch memory, one for a predetermined percussive sound, and the other for a predetermined sustained sound.

The Software Version

Unlike the Kontakt instruments that are its competition (see box) Cherry's emulation is generated by DSP rather than samples, so perhaps this is why its polyphony is limited to 32 voices. But there are many other differences between it and the original. For example, its Balancer acts as a key slope centred on C2, diminishing the level on one side while emphasising the other. It's a valid approach, but it's not how the original functioned. Also (for obvious reasons) any damper pedal connected to your MIDI controller operates across the whole width of the soft synth, whereas the original had three pedals — one for the lower 36 notes alone, and two that affected the whole keyboard.

Moving on, Bright-Mellow is now a continuous control rather than a two-position switch, and it filters the whole keyboard range. Again, this is valid, if not authentic. However, it seemed to me that its performance was not quite right, so I asked Cherry Audio about this and they told me, "Our implementation of Bright/Mellow is in error, and we're currently wrapping up work on an update to address it." I commend them on their lack of bullshit; by the time that you read this, all should be well. Another error lies in the release time, which is rather too short, so I hope that this will also be addressed.

Cherry Audio
Novachord & Solovox
$39

PROS
- Two unusual soft synths that might send you down unexpected musical paths.
- The Novachord is capable of creating some gorgeous strings, pads and choral sounds.
- Cherry's MIDI Learn system extends both instruments considerably.
- They're not going to break the bank.

CONS
- They're not wholly accurate recreations of their inspirations — particularly in the case of the Solovox.
- Velocity sensitivity would be nice.
- They will prove to be too limited and limiting for many players.

SUMMARY
Cherry's latest offering comprises a pair of limited but unusual soft synths that drag two of the earliest attempts at synthesis into the modern world. They're not for everyone and I doubt that they will be at the centre of your music creation, but they could add interesting flavours that you won't find elsewhere.

Turning to the modulation, the soft synth employs six LFOs oscillating at very slightly different frequencies to emulate the Novachord's vibrato but, instead of having fixed routings, these are assigned as you play, which the developers feel offers a more natural quality to the sound. In addition, the modulation generator offers two waveforms. This is because there was a change in the waveshape from square-ish to triangle-ish during the Novachord's production run. Other improvements include a continuous volume control and the addition of a room/hall reverb that adds a huge amount to the untreated sound. There's also a simple limiter but, with correct programming, you should never need this.

So... how accurate does it sound? Perhaps the most apposite response to this is, "Why does that matter? You're never going to get the chance to conduct an A/B test!" But, if pressed, I would suggest that, while it has the soul of the Novachord, it's a little too polite to be a perfect emulation, perhaps because the well-behaved oscillators, filters and modulators don't capture the instabilities of the original. Whether this is a fair comparison (after all, the only referents are roughly 80 years old!) and whether it's a shortcoming or a blessing is for you to decide. But if there's a tangible disappointment, it's that Cherry declined the temptation to add velocity sensitivity. This may seem the purists' approach, but the prototype of the original was velocity-sensitive, and this feature was only removed when Hammond discovered that it was too expensive to implement on production models. I hope that the developers will add this to the list of MIDI control sources in a future update.

But what of the sound itself? In truth, it can be gorgeous. Played through a tube saturation emulator and then your favourite selection of choruses, delays and reverbs, the soft synth shares much of the quality so beautifully captured by Dani Wilson's samples of #346. Patch it and play it like a cheap organ and it will sound like one but, with a bit of careful tweaking and sympathetic playing, solo strings and ensembles just pour out of it, as can ethereal voices that make my Roland VP-330 throw jealous hissy-fits. Layer it using two or more instances and swoon. Of course, the Novachord architecture is capable of more than that but its palette is, by modern standards, very constrained so don't expect too much of it and remember that a small selection of beautiful sounds is worth more than a huge library of mediocrity.

Coffee & Mints

Bundled with the Novachord, Cherry's Solovox looks and feels nothing like its namesake, which was similar in design to a Clavioline. The differences are so significant that I contacted the company to ask whether the two are related at all. I was told, "We felt the original Solovox's interface was very limited and that, with its switches underneath the keys, it would've been difficult to represent. We also felt that a straight emulation would not be interesting as a plug-in. So we decided to blend its speaker, vacuum tubes and fundamental controls into an updated interface that is more visually interesting, and we implemented the controls as continuously variable knobs to provide much more flexibility than the fixed switches of the original."

Its voice generation starts with a single oscillator offering a button ('Mute' — as in a brass instrument's mute, not silence) to switch between a sawtooth-ier or a square-wave-ier tone, and four octaves »

Cherry Audio have completely reimagined the Solovox's front panel.

In common with other Cherry soft synths, the Solovox allows you to assign MIDI controllers to voice parameters, each with minimum and maximum values and selectable response curves. Assignments can be global or on a per-patch basis. You can even direct a single controller to multiple destinations. Make it sing!

» of the octave-divided waveform, selectable in any combination. The signal then passes through five parallel audio paths, with a low-passed version (deep tone), an unadulterated version (full tone), two band-pass filtered versions with adjustable peak frequencies (first and second voices), and a high-pass filtered version (brilliant). This time, the loudness contour generator is an ASR offering variable attack (percussion/singing, not to be confused with Korg's different use of the same terms) and release (short/long). A fixed vibrato is also provided but there's no way to adjust the depth of this, even when using MIDI CCs or other controllers; it's either on or off. Finally, there's an exponential glide with adjustable rate, and a basic reverb with no controls other than depth. And that's yer lot, baby!

In use, it sounds like what it is — a basic and well-behaved monosynth. However, I've never heard a well-behaved instrument of the Clavioline family. I've owned three, and all of them were brash, harsh, aggressive... choose your adjective. There's something about the Cherry Solovox that's too damn nice, and a more peaky response with some well-analysed and recreated grit and distortion in the signal path would have brought it closer to the original, especially an original suffering from decades of electronic degradation. Nonetheless, it allows you to recreate sounds redolent of the mid-20th Century, and its limitations force you into the same territory as before, whether recreating 80-year-old imitations of flutes and clarinets or playing 'Telstar'. Whether that appeals or not is, as always, a matter of taste.

Final Thoughts (For Now)

Cherry Audio have made a name for themselves by creating imitative soft synths and selling them at affordable prices. Although it's questionable whether there was a need for yet another 'soft' Minimoog or ARP 2600, many of their products have been a little more leftfield and have represented excellent value for money even though one or two have fallen a tad short when compared directly with their inspirations. But this isn't a criticism that will be levelled easily against either of the instruments reviewed here because it's rumoured that as few as five fully functional examples of the Novachord now exist, and the Solovox fares little better.

So kudos to Cherry Audio for making these available to us in modern form, albeit with many changes from the original architectures and controls. Speaking of which... both soft synths also include Cherry's MIDI Learn system, which means that you can map aftertouch, any MIDI CC, MMC, or key on your QWERTY keyboard to any number of controls on their panels with your choice of range, polarity and curve for each. Within their limitations, this makes both instruments expressive in ways that Laurens Hammond may never have dreamed possible. Inevitably, these changes and extensions may not appeal to purists. On the other hand, there may be too few additions to appeal to potential owners who like their soft synths to include the dining room table as well as the kitchen sink. But if you fall between these extremes and have a penchant for something a different from the likes of the myriad Minimoogs, Odysseys, Prophets and Jupes that dominate the pantheon of imitative soft synths, they're worth a look. I suspect that the Novachord is the one that will persuade you to hand over your cash and that the Solovox will be viewed as a bonus, but who knows... some players might love it. Either way, let's party like it's 1939.

Many thanks to Dani Wilson at Hideaway Studios for her help with this review.

$ $39
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The Alternatives

Many years ago I was contacted by Dani Wilson, a vintage synthesizer specialist who had purchased a Novachord from synth guru Marc Doty and had it shipped to the UK. (The story behind this instrument — serial number 346 — and its loving restoration is told in detail at www.novachord.co.uk.) Not long after receiving it, Dani sent me a selection of samples that blew me away. There was something magical about them, especially when treated with modern effects, and Dani subsequently partnered with Steve Howell at Hollow Sun to create a rather excellent Novachord Kontakt instrument (*SOS* July 2010). Competition for this arrived not long after in the form of another sample-based recreation from Soniccouture (*SOS* October 2010), which is based upon Phil Cirocco's instrument in the US. Samples or DSP? I'll leave you to decide.

Image courtesy of D.A.Wilson, Hideaway Studio

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KMI K-Board-C & QuNexus RED

Controller Keyboards

KMI's latest keyboards offer MPE control in compact and colourful packages.

KMI K-Board-C & QuNexus RED
$119 & $199

PROS
- Both keyboards are light, super-portable MPE-capable controllers.
- The QuNexus packs a rich selection of performance features and interfacing into its compact form.

CONS
- Keys are pressure-sensitive only and don't move.
- Position sensing (tilt) for pitch can be a bit hit-and-miss.
- QuNexus clocked notes don't always seem to hit the beat.

SUMMARY
The K-Board is a bare-bones, MPE-capable MIDI mini-keyboard in a form factor that makes it super-portable. The QuNexus has the same form factor but includes CV/gate and MIDI expander connections and includes a multitrack arpeggiator and step sequencer.

NICK ROTHWELL

For quite some years now, Keith McMillen Instruments (KMI) have been producing an assortment of MIDI and audio devices with LED-backlit pressure-sensitive pads. Here it's the turn of two small, affordable keyboard controllers: the K-Board-C and the QuNexus RED, each the latest revision of earlier K-Board and QuNexus products.

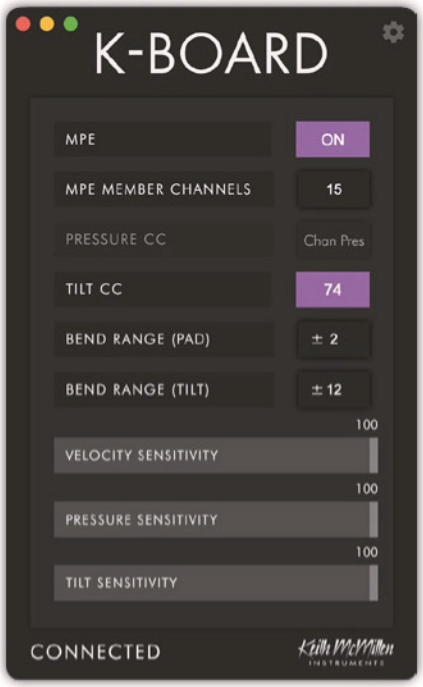
The K-Board and QuNexus are pretty much identical in terms of shape, size, construction and features. The QuNexus sports some extra sockets that the K-Board is missing, and much more labelling on the top panel. The K-Board is available in a variety of colours, but thankfully, my review unit was a rather conservative slightly metallic-looking grey. The case is plastic, though it seems pretty solid and tough. The QuNexus RED is, as its name suggests, bright red in colour making it look more toy-like than the K-Board, even though it's the more expensive and capable model.

Each device sports a two-octave C-to-C keyboard of mini-key pads, all white-ish in colour; these light up when played, the naturals in white and the sharps/flats in blue (which is the closest that modern technology comes to lighting up in black). The left-hand side of the playing surface provides controls for octave shift, pitch-bend and various programming options, such as selective enabling of velocity, aftertouch and key position or 'tilt'. Both devices are USB powered and connect to a host via a USB-C socket on the left-hand edge.

K-Board Meeting

We'll start with the K-Board, which is the simpler of the two models. The 'keys' don't actually move, except for a very slight compression under pressure. Despite the lack of physical movement, the keys do generate MIDI velocity which feels pretty responsive and authentic. Channel aftertouch is supported, but since the keys operate by pressure, playing a key with high 'velocity' will trigger a note with considerable aftertouch at its onset.

Pitch-bend is less than ideal. There's a small dedicated pitch-bend pad providing fiddly and not very accurate pitch control; I think if the physical pad were longer, it might work better. Rounding off the controls, there's a momentary Sustain button, and a Toggle button, which turns the keys into



Settings in the K-Board Editor.

note toggles, useful for holding chords or setting up drones. There's a pair of octave up/down buttons, but they are rather small and rather too close to the bend pad and bottom C key, so it's a bit too easy to accidentally trigger something when switching octave, or to accidentally switch octave while playing.

To unlock the full expressive power of the K-Board, it needs to be switched into MPE mode in the dedicated editor app (and, of course, you need an MPE-capable DAW and/or instrument). Polyphonic pressure works fine, with the proviso that notes trigger with some pressure already applied, as noted earlier, so fine control is difficult — some kind of pressure 'dead zone' would be useful here. MPE Glide (polyphonic pitch control) is something of a mixed bag: the K-Board and QuNexus don't have side-to-side sensing, and so have to use front-to-back 'tilt' instead. Pitch transmission seems to be based on absolute finger position on the key (or possibly difference in pressure between top and bottom), so triggering a note with no pitch modification is pretty much impossible, as is fine-tuning the pitch once triggered: some kind of dead zone for pitch would have been useful here as well. 'Tilt' mapped to pressure is a lot more practical: more so than using actual key pressure, which works best for generating note velocity. If you're not put off by the rather unruly pitch control, or don't need it, the K-Board is eminently playable and expressive.

Thank QuNexus

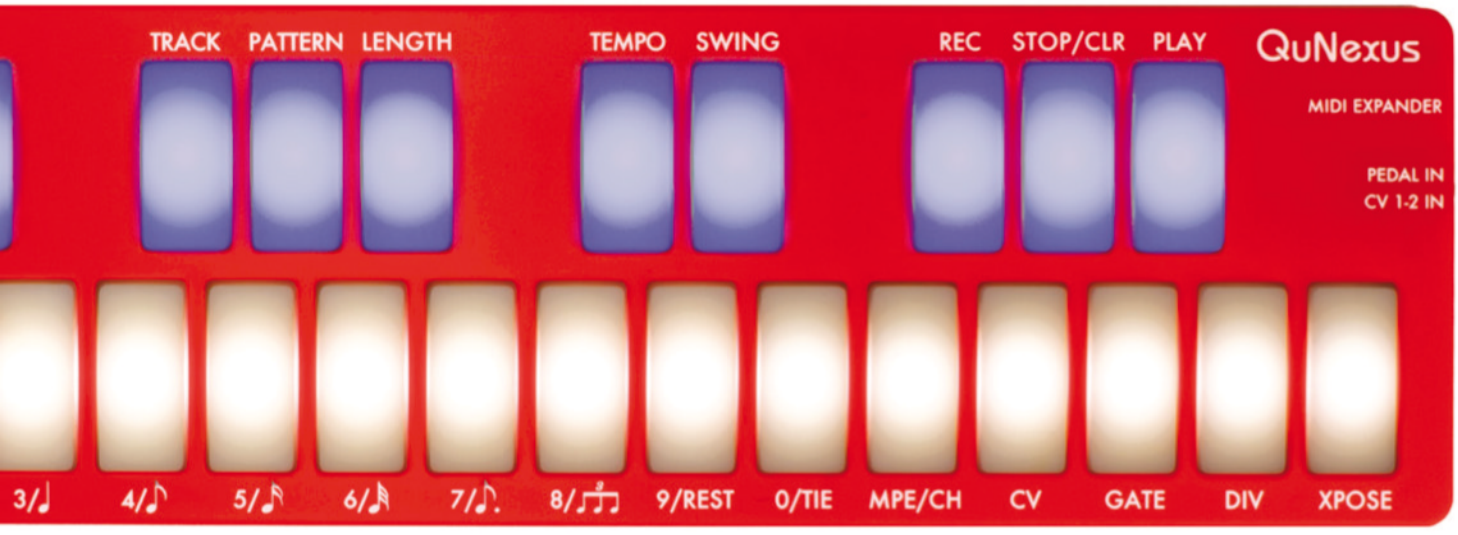
Compared to the K-Board, the QuNexus RED is a much more sophisticated beast. The word 'nexus' in the name hints at this, as do the labels on the top panel and the sockets on the sides of the case, primarily dedicated to CV/pitch interfacing (which, unfortunately, I was not equipped to test). The editing app is also a lot more detailed. The QuNexus can do everything that the K-Board can, and although it's missing a sustain button an actual sustain pedal can be used instead.

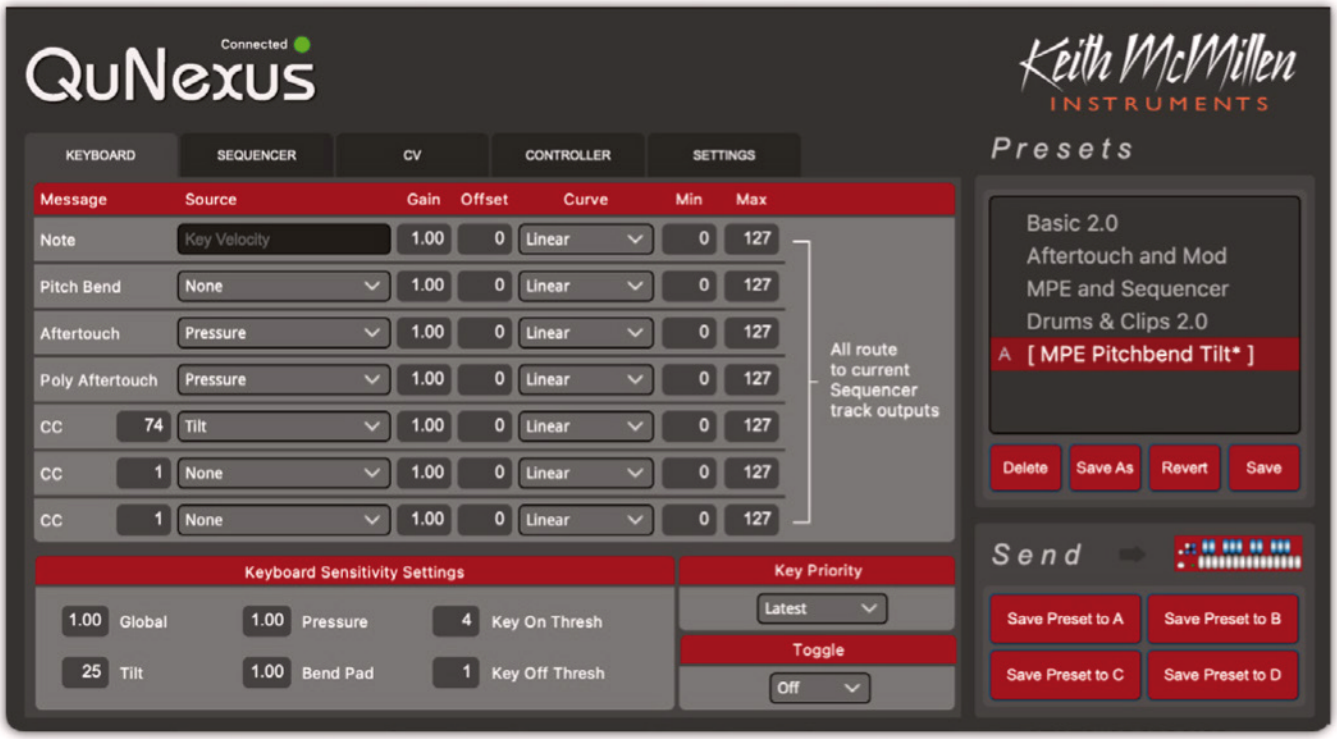
The QuNexus sports three internal MIDI tracks, and at any time the keyboard is attached to one of them. Each track can have its MIDI messages routed to the USB port, like a standard controller, or to its 'expander' port, or both. The expander port is a mini-USB socket on the right-hand edge of the device, and the QuNexus comes with an adaptor to turn this into a 5-pin MIDI output. Hence, the QuNexus can operate as a bare MIDI controller without a computer (though it still needs USB power). Or, should you wish, you could route one track to USB (into a DAW) and another purely to the physical MIDI output, with further options to route to/from CV and gate. Track 1 can operate in MPE mode, in which case, as per the MPE specification, it allocates a number of MIDI channels to support polyphonic expression. Tracks 2 and 3 are non-MPE and can each use a spare remaining channel.

At this point I feel I should mention the QuNexus user interface, which could be described as inscrutable. Enter edit mode by holding Shift and you're faced with various patterns of flashing note keys to tell you what's going on, and assorted key-press sequences to change things. After a bit of practice it all starts to make sense: there are the helpful labels on the device, and muscle memory kicks in pretty soon. But if you really want a clear visual indication of every detail of what's going on, the QuNexus editor app shows everything on screen and stays synchronised with the device.

There's an arpeggiator and a step sequencer, per track, and a global clock for timing, with play, stop and record controls. Each track is in one of two modes: arpeggiating or sequencing. When the device's global clock is stopped, the keyboard just plays notes to its chosen destination. When playback is started, the three tracks start arpeggiating or sequence playing in time and in parallel. It's not generally possible »

The QuNexus RED.





Keyboard settings in the QuNexus Editor.

» to have a sequencer or arpeggiator running on some of the tracks and not others: either the clock is running and they're all active, or the clock is stopped and they are not. Only a track in sequence mode with no sequence data recorded will behave like a dumb keyboard.

With the device's transport stopped, notes are entered into the sequencer step by step after pressing Rec. You can also go back into Rec mode to add more notes. There's also a real-time mode: with the transport running, hitting Rec will let you record what you play, and will punch in over the current sequence data in the style of a looper. Eight sequencer patterns can be saved per preset.

The arpeggiator supports a standard selection of patterns (including both random and Brownian motion, the latter being particularly good for generating meandering melody lines) and a repeat range of up to four octaves. It has not one hold mode but three: the first mode holds or sustains notes until all keys are released, the second turns keys into toggle switches for notes, and the third allows active notes to be 'replayed' to change velocity or expression settings. The step sequencer is also subject to pattern selection and octave repeat, a useful source of musical ideas.

MIDI clock quantising seemed a little hit and miss: arpeggios appeared to start synchronised to the first note played, not the downbeat of the clock, making DAW synchronisation tricky, and it wasn't clear how sequence playback was aligned. Letting the QuNexus be master clock and the DAW follow it was much more predictable.

QuNexus Editor

Although a lot can be achieved using the keyboard and button shortcuts on the QuNexus itself, for really detailed editing you have to fire up the editor. The editor window is a bit small and fiddly, and the overall design looks rather dated: this could do with an update and a design refresh.

One important function of the editor is that it manages and saves presets. Any of the named presets can be loaded into any of the four preset slots on the device.

The editor allows for more detailed configuration such as response curves, key priority, CV/gate configuration, and a totally distinct mode called 'controller', where each key can be given its own static MIDI output settings: as well as note pitch and velocity, a key can be a toggle, with controller output as well. Keys can be taken out of the standard 'keyboard layer' and made part of this controller layer instead — or in addition. These controller mappings are per preset. This feature is probably most useful if you want to partition off a group of keys on the QuNexus to make MIDI bindings to control an instrument or effect such as a looper.

Conclusions

There's a lot to like about the K-Board and the QuNexus. They are robust and portable, and while the K-Board provides a no-nonsense set of features for basic MPE keyboard control, the QuNexus is a sophisticated controller with arpeggiator, sequencer and a varied selection of I/O options. The K-Board is hard to fault as a solid bare-bones controller, if you can get along with the fact that the keys are pressure pads with little movement or physical feedback, and are forgiving when it comes to pitch control. The QuNexus is a more complex offering, and while its keyboard-based interface is somewhat obscure, I got used to it quickly. I found some features to be a bit temperamental, but it worked well as an ideas generator and was fun to use. I think my preference leans toward the QuNexus for the sequencing and arpeggiation features, although I'd have liked a bit more timing precision when using them. In any case, the form factor and construction makes both models useful to have in your studio toolkit.

$ K-Board-C $119, QuNexus RED $199
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AJH Synth MiniMod Keyz

Modular Synthesizer

AJH Synth's first dedicated system reimagines what the Minimoog might be.

GORDON REID

The concept of a small, pre-configured modular synthesizer goes all the way back to 1971, when Moog Music released the Model 10, the cabinet of which contained all of the modules needed for what we might now consider to be conventional analogue subtractive synthesis. Unfortunately, the Model 10 was still quite bulky, and its keyboard was supplied as a separate unit, which was fine in the studio but much less convenient for gigging musicians. A better solution had already appeared when Moog and their competitors ARP had chosen a selection of functions — this many oscillators, that many filters, and so on — and integrated them with keyboards and other performance controls within single cases. These instruments may have been less flexible, but they were much more usable so, by the mid-1970s, modular synthesis had largely become the preserve of the wealthy and the fanatical. But two decades later, affordable Eurorack modules started to appear and the world was soon awash with them. It didn't take long for manufacturers to realise that several of these smaller and lighter devices could be mounted within a specially designed keyboard to create a self-contained instrument that combined the flexibility of a modular synth with the convenience of an integrated one. I suspect that the AS Sorceror was the first of these, followed by similar offerings from other companies who, like Analogue Systems, supplied them as empty frames ready to receive the modules of your choice. I understand why; Eurorack enthusiasts love to tweak and experiment with the architectures of their systems, removing existing modules and replacing them with others to achieve a desired outcome. (Or, if we're honest, just to have a bit of fun because something new appeals.) But I've often felt that a pre-configured Eurorack keyboard/synth targeting a specific outcome — say, a recreation of a small modular Moog — would be a rather neat idea, especially if it

added performance enhancements such as velocity sensitivity and aftertouch. And here it is.

In The Box

The primary modules in the MiniMod Keyz allow you to recreate the architecture of a Minimoog. They comprise three VCOs (reviewed in *Sound On Sound* June 2016), a Transistor Ladder VCF (*SOS* July 2016), and a VCA. These are accompanied by a Dual Contour Generator and a Glide+Noise module. The system also includes three modules that offer facilities not found on the Minimoog. The first is called the Low Fat Filter, which, despite its ambiguous name, is a high-pass filter offering 6dB/oct and 12dB/oct cutoff slopes. The next is the Ring SM ring modulator/sub-bass generator/mixer, which was described as "a winner!" here in December 2016. Finally, to overcome a significant limitation in the Minimoog architecture, there's a dedicated LFO, which, with its 16 initial waveforms, waveshaping, CV control, tap tempo and more, goes way beyond using VCO3 as a modulator!

All of these are housed in a 49-key, 112HP Cre8audio NiftyKEYZ controller/rack (*SOS* October 2022). The name is crap but, happily, almost everything else about it is not. Far from being a simple monophonic generator of pitch CVs and gates, it's a polyphonic keyboard that generates up to four CV+gate pairs in two zones, with selectable single- and multi-triggering modes. But it's not just a passive host for the modules mounted within it. There's a latchable four-octave arpeggiator with a 32-step sequencer mode, a further LFO, master and slave clock modes, and chord memory. Performance controls comprise pitch-bend and modulation wheels, plus transpose and octave shift functions. But best of all, it's both velocity and pressure sensitive. Unfortunately, there's no space to revisit it in depth here; there's lots to discuss but, if you want to know more, you'll have to refer to the previous review.

In Use

Given that almost all of its components have been reviewed in isolation, the question we have to ask is how well the MiniMod Keyz works as a system. After all, you may like eggs, bacon, strawberry yoghurt, chillies and jellybabies, but would you like them served as an omelette? To test this, I started by setting it up to perform as a Minimoog. I connected three cables from the oscillator outputs to the main filter inputs, one from the filter output to the amplifier input, and another from the amplifier to the inputs of the outputs (if you see what I mean). I then patched cables from the dual contour generators to appropriate CV inputs on each of the filter and VCA. Now, with the NiftyKEYZ in one-voice mode, I connected its first three CV outputs to the 1V/oct CV inputs of the oscillators, and voice 1's gate to the appropriate input on the contour generator module. (I could have connected a single cable from CV1 to the Glide+Noise module, which is itself connected to the 1V/oct CV inputs of all three oscillators via an internal bus but, to test the consistency of the CVs and enable me to jump through other hoops later in my tests, I patched the three oscillators separately.) Next, I connected the fourth keyboard CV to the filter's 1V/oct CV input to create key tracking, and the triangle wave output from the third oscillator (via a multiple) to CV inputs on the first and second oscillators to generate modulation.

By this point, many of the cables that I had used were trailing across the keyboard so I set about repatching it to ensure that the cables and my fingers weren't occupying the same space in ways that they shouldn't oughta. This problem is far from unique to the MiniMod Keyz, but it's less of an issue on my Sorceror (which, to be fair, is a much larger, heavier and, if we're honest, more expensive piece of kit) »

AJH Synth MiniMod Keyz

$3499

PROS

- Who can complain about a small, affordable but flexible modular Moog? Well, not a Moog, but you know what I mean.
- Forty-nine keys, velocity, aftertouch, four-voice CV+gate architecture and more... there's a lot here to like.
- Its sound is deep inside Moog territory.
- It looks great.

CONS

- It would benefit from more mixers and multiples.
- Unless carefully placed, the patch cables can interfere with playing.
- I wouldn't recommend that you use the NiftyKEYZ as a MIDI controller keyboard.
- It has an external PSU with a short cable and no stress relief.

SUMMARY

Since the 1970s, players have dreamed of a modular Minimoog with extended voicing and performance capabilities. Within the limits of emulative technology, that's what we have here. It's a powerful and flexible system that can sound excellent.

The Rear Panel

The NiftyKEYZ's rear panel is sparse but, given that most of its connectivity is provided on top where it's of most use, that's fine. To the left lie two unbalanced quarter-inch audio outputs and an associated headphone output while, to the right, you'll find two control pedal inputs for expression and sustain. Between these lie 5-pin MIDI in and combined out/thru sockets, plus USB for MIDI in/out. Unfortunately, power is provided by an external PSU via a barrel socket with no associated strain relief hook. What's more, the PSU cable is remarkably short. You know what I'm going to say, don't you? Given all of the thought lavished on the design in other areas, this is a disappointing lapse. The NiftyKEYZ — and, by extension, the MiniMod Keyz — deserves an internal PSU and an IEC socket.

As well as the actual keyboard and case, the NiftyKEYZ brings plenty of other things to the party, including an arpeggiator.

» because there's a slab of wood between the keyboard and the modules. But it illustrates why other patching systems have enjoyed popularity at various times. ARP favoured switch matrices while EMS opted for a pin matrix — a system recently resurrected by Arturia and Erica Synths in electronic form. Not only do these approaches dispense with patch cables, they also allow you to connect a single source to multiple destinations and multiple sources to a single destination without additional mixers and multiples. But 3.5mm cables rule the Eurorack world, which is a shame because the MiniMod Keyz looks great without them, and you can get to the knobs and switches more easily!

No doubt you're now wondering how it sounded. Given everything that we already know about AJH modules, it came as no surprise to discover that it sounded much like a Moog synthesizer. Was I going to compare it with a genuine Minimoog? Absolutely not because, having patched it in what I viewed was quite a basic fashion, my aim was to take it further into esoteric sonic territory. So I stopped trying to recreate the 1970s and started to approach the system as an instrument in its own right. To do so, I first connected the velocity CV to one of the VCA's CV inputs and the aftertouch CV to the Level input of the LFO. I also allowed myself to take advantage of the system's ring modulation, sub-octave generators, audio/CV mixer and high-pass filter, which hugely extended the sound creation potential. Unfortunately, like almost all modular synths, the MiniMod Keyz includes too few mixers, multiples and independent VCAs. Due to the paucity of the second of these, I was unable to make all three oscillators respond simultaneously to the LFO unless I resorted to an external multiple, but the system packs a lot of synthesis into a small space and, with a bit of delay and reverb, the results could be gorgeous. Whether you prefer to make synths go 'bleep' or are a fully paid-up member of the widdly-widdly brigade, there's a lot of potential here.

Because of the way that I had patched the pitch CVs, I was also able to experiment with the system as a three-voice paraphonic synth. But I had limited success with this, so I didn't pursue it for long; the strength of the MiniMod Keyz lies in sounding and responding like an enhanced but monophonic modular Moog.

Putting it through its paces wasn't without its interesting moments. On one day, the NiftyKEYZ glitched (or maybe overheated — it was a ridiculously hot afternoon) and its pitch CVs became unreliable. On another occasion, pitch CVs could be 'left behind' when I played rapidly. I spoke to Allan Hall at AJH Synth about this and, while we never got to the bottom of it, he undertook to add quality control checks for this. That's an excellent response from the manufacturer.

My other reservation arose when I used the NiftyKEYZ as a MIDI controller, mixing the AJH sounds with external ones. Its keyboard felt slow, spongy and offered an uneven response in a way that it didn't when playing the installed modules alone. I know that the review unit is AJH Synth's heavily used test-bed, so I can offer it a little latitude, but not much. MIDI control isn't its *raison d'être* but, whether this issue is a consequence of the keybed, the processing or both, I believe that the NiftyKEYZ should feel and respond better.

Conclusion

So the final question has to be, is the system worth the price? A quick calculation shows that the cost of the individual AJH modules is currently £2315. Add to this the £529 that the NiftyKEYZ will set you back, and the total comes to £2844. Consequently, the system price of £2795 represents a small saving, and you get the benefit of an assembled and tested system. So let's ask a different question. In an instrument of this nature, should you constrain yourself to a set of modules from a single manufacturer? I think that the answer to this is simple. If you want to build your own synthesizer architecture, it's not the correct approach. But if you're after 'the Moog sound', the modules in the MiniMod Keyz are designed to work together to provide this to a high degree of authenticity. It's not all-encompassing — no Eurorack system of this width can be — but, if you like that sound (and, let's be honest, almost all of us do) I don't think that you'll find the MiniMod Keyz to be unreasonably priced or lacking in creative possibilities. There are players who are going to love it.

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Alice 538R

Stereo Optical Compressor

If you're a fan of Ted Fletcher's classic optical compressor designs, this one shouldn't disappoint.

MATT HOUGHTON

I've always had a soft spot for Ted Fletcher's optical compressors. His JoeMeek SC2 became something of a classic 'character' compressor for drums, of course, though it deserves praise for more than that one 'trick'. Most of Ted's compressor designs over the years have been based to some extent on the same circuit — if it ain't broke, don't fix it — and I've owned and used quite a few of his compressors and channel strips. I've enjoyed them all: classy, yet full of character, and with a little less hiss in the later models than the earliest. So when he sent me his latest creation, the Alice 538R stereo optical compressor, I was very happy to take a look.

Overview

The 1U 538R isn't something you'll overlook nestling amongst the other gear in your 19-inch rack: the chassis, which extends 15cm (six inches) behind the rack ears, is painted a distinctive... I'm not quite sure. I could call it 'muted magenta', maybe or 'heather meets lilac', but certainly it seems a bit more vibrant in the flesh than most images I've seen suggest.

Alice 538R
$838

PROS
- Classic Ted Fletcher optical compression.
- Versatile.
- External side-chain input.
- M-S Width control.

CONS
- Small turns can do big things!

SUMMARY
Though built using modern methods, the Alice 538R delivers a compression and sonic character that will be familiar to anyone who's used Ted Fletcher's processors before — and very appealing it can be, too!

With the white legends, the warm glow of the meters and the purple power LED, it all comes together pretty nicely.

On the front panel, to the right of a power-present LED, two moving-coil meters have a small toggle switch between indicating the left and right channel output levels or the summed stereo output (left) and gain reduction (right). Another toggle instructs the compressor to react to the internal or external side-chain signal.

The main controls comprise two clusters of knobs, which flank a Char (character) toggle, of which more later. To the right of the switch, the Input knob (marked 1-10) is a pot to set the level before it hits the compression circuit. Between this and the meters, it's very easy to set a signal level so that it's in the right ballpark for compression. The Width control appears to be an M-S balance pot, and ranges from mono (fully anticlockwise) to 150 percent (fully clockwise), with the balance unaffected at 12 o'clock (detented). The third knob in this trio (again marked 0-10) sets the Output level. It comes after the meters in the signal path, and is there to set an appropriate level to feed the next device in your chain.

To the left of the switch are four knobs, all continuous pots. Comp sets the amount of compression, by raising the signal going into the compression circuit. Attack runs from Slow (fully anticlockwise) to Fast (clockwise), while Release has the fastest time fully anticlockwise, and Ratio is marked 1.2:1 (very subtle) to 10:1 (effectively limiting). The Char switch toggles between Norm and Dark positions, but the manual's a little coy about what's actually happening: a single mention informs us that this "artistically alters the release timing profile". Finally, a blue On (bypass) button, which some may find a touch too bright, lights when pressed in, in which case the compressor is in circuit; the out (unlit) position is a hard-wired bypass.

Inside, the construction is neat and tidy, and almost all components are mounted on a single PCB, with most (though not quite all) being surface-mount devices. Gain reduction is performed by a pair of Alice-badged optocells — the higher the signal, the brighter an LED inside this sealed cell glows, increasing the resistance of its light-dependent resistor. The internal power supply can accept 85-285 V AC mains at 50/60 Hz through the rear-panel IEC inlet, and creates ±15V power rails for the internal circuitry. Also on the rear are four XLRs for the analogue I/O and a quarter-inch TRS jack side-chain input. The inputs and outputs are transformer balanced, and the shielded transformers (manufactured by OEP, a Carnhill sub-brand) present input and output impedances of 10kΩ and 100Ω, respectively. The frequency response is within 1dB of flat across the audible band and various other decent-if-not-groundbreaking specs are detailed in the manual. Suffice it to say

that with no processing taking place this thing sounds subjectively clean and quiet, and the more you compress the more character is added in terms of harmonic distortion.

As with all optical compressors, the attack, release and even the ratio vary somewhat with the signal level. But an interesting aspect of Ted's stereo compressors made in the last couple of decades is that the compression is performed not on the left and right channels of a stereo signal but on the Mid and Sides. This eliminates the problems of channel imbalances owing to differences in the dynamic response of opto cells. A potential downside is that it means you cannot used both channels as mono compressors (though you can use one channel in mono if you wish). A bonus is that since M-S is an intrinsic feature of the compressor design, Ted has taken the opportunity to present the user with the M-S balance (Width) control.

The M-S-based Width control is an unusual but useful bonus, and a result of the approach Ted Fletcher uses to prevent discrepancies between two channels in his stereo optical compressor designs.

In Use

When the 538R arrived, I was working on a rock mix featuring a good range of instruments, including multitrack drums, double-tracked male lead vocals and stacked backing vocals, electric bass, several electric guitars and a piano. So I inserted the 538R on the master bus and started soloing different sources, just to get a feel for what it had to offer. Setting levels was easy: I could just turn Comp right down, turn Input up until I saw a healthy level on the meters, then set the Output level while watching my audio interface's meters. Flipping the meters to show gain reduction, I found the controls pretty intuitive, but because of the wide range they offer they could seem a tad sensitive — on some sources where I wanted less 'character' I had to spend a little more time than I'd have liked optimising the settings. That said, I could always set things as I wanted.

On a drum bus, I coaxed a characterful, explosive, pumping sound out of this compressor very quickly — the sort of effect that's great in parallel compression. There's no wet/dry mix control here, which some will miss, though I can't say I did, as I generally prefer to put the parallel processing on its own channel, where I can EQ it and leave the dry level unchanged. Spending more time with the drums, I soon got a feel for the lightness of touch required to achieve more delicate compression; the slightest twist is often all that's needed, but the knurled metal knobs make this easier than it sounds.

Of course, it's not all about drums. I often find that bass responds well to optical compression (and *vice versa*!) and for a while I had a Fletcher-designed channel strip as my main bass input channel. The 538R didn't disappoint. It did a decent job of controlling the peaks and making bass parts sound that bit fuller, without being too obvious or losing detail. I could achieve a lovely, almost tape-like smoothing/gluing effect on the guitar bus, again without things turning to mush. Generally, though, you'll find that the more delicate or exposed the source and the subtler the result you seek, the fussier you need to be with the controls. It was capable of a lovely result on lead vocals, for example, but the controls tend to interact a bit (largely, I think, because the ratio and time constants vary with the level/amount of compression) and I found I wanted to spend a fair bit of time massaging the Comp, Input and Output knobs to achieve the best result.

The Width control is such a useful feature to have on board, and not just for the drum or mix busses. It can be very effective at bringing up the room ambience or reverb (or for pulling the centre into sharper focus) in any stereo signal, from a piano recording to looped samples, and I enjoyed the immediacy of being able to make those decisions as I tweaked the compressor. By way of example, I loved using the 538R to control some stacked backing vocals that had been panned across the 'stage' and fed through a stereo reverb. Feeding both the vocals and their reverb through this thing, the combination of gain reduction and width control smoothed things out beautifully, adding a dash of thickness and bringing up details, while also 'tucking' the parts in place around the lead vocal. When trying the same thing with plug-ins I could reach the same destination, but nowhere near as quickly.

I get why the Char settings are labelled Normal and Dark: the latter results in what I perceived as a slight dulling of the top end, with perhaps a tiny drop in level too. But really it's a subtle 'flick and listen' thing, and I almost always preferred it in the Normal position. I can't say I'd miss it were it not here, but it would be harsh to criticise the presence of a feature!

Verdict

That Ted Fletcher's compressors aren't talked about more these days has long puzzled me. I have lots of other nice hardware and software compressors available in my studio, but his designs always seem to offer me something different and I've enjoyed every one I've used. Perhaps it's because they've been around for so long and haven't been backed with a big marketing spend? Either way, the Alice 538R is true to its lineage, and it's well worth checking out. The controls may sometimes seem a tad sensitive, but you soon get a feel for it, and when you do, you'll almost certainly enjoy what you hear.

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Apple Logic Pro For iPad

Digital Audio Workstation Software

With their iPad version of Logic Pro, have Apple succeeded in implementing a touch-first user interface without sacrificing pro features?

MARK WHERRY

Large parts of Logic's underlying audio engine have been running on iPads since 2011, when an iOS a version of GarageBand was introduced. And what Apple so deftly demonstrated with GarageBand was how to create a user interface tailored to the delicate, yet imprecise, caresses of our fingers. Rather than simply port the Mac version of GarageBand, Apple reimagined it, creating an app that was both recognisably GarageBand yet clearly defined for the iPad.

Now Logic Pro itself has made the move; and to succeed, it too needs to have a reason to exist on the iPad, embracing the interaction models unique to this device, such as touch-first and Pencil-based input. Logic Pro For iPad can't simply be an imitation. It must be a reinvention that retains the essence of what makes the application attractive for musicians. Is it?

A Logical Start

The Logic Pro experience on the iPad starts with the Project Browser, which is based on the standard iPadOS Document Browser and allows you to create new Projects and manage existing ones. These can be stored on your iPad, iCloud (or other cloud-based storage), or a server, making it possible to access any Macs that might be attached to your local network. A collection of interactive lessons explains topics such as recording vocals or software instruments, the Step Sequencer, and so on, complementing the well-written and usefully organised online help that runs to 959 pages in PDF form.

New Projects can be based around the typical Tracks area or a Live Loops grid, and the New Project Setting allows you to define an initial tempo, time signature, key signature and sample rate. The iPad's internal clock supports 44.1, 48, 88.2 and 96 kHz, with 48kHz being the sensible default.

Track types have been tidied into four categories: MIDI, Audio, Pattern and Drummer. Support for hardware MIDI instruments is included courtesy of the External Instrument plug-in, and MIDI tracks can trigger either this plug-in or a software instrument.

If you'd prefer a new track to be based on a pre-existing sound, you can tap the Browser button. This opens a new view consolidating the traditional Library and Loop Browser views from the Mac version into a single catalogue of instrument and audio patches, Apple Loops, samples, presets, and patterns, with handy shortcuts for accessing recently used items or downloading additional Sound Library content. With so much content on offer — around 15GB — a new filter view offers advanced options making it easy to find exactly what you're looking for, and the way in which Presets are managed alongside Audio and Instrument Patches is also rather helpful. Presets store the state of the controls within a single plug-in, whereas Instrument and Audio Patches encapsulate the configuration of plug-ins and routing within an Instrument or Audio track.

An App With A View

Usable in either landscape or portrait orientations, Logic Pro For iPad's interface is divided into several distinct areas, beginning with the Control Bar running along the top of the interface. This can be customised to show various transport-related controls, displays and modes, whilst also providing access to settings or the Project Browser.

The main part of the interface shows either the Tracks area or the Live Loops grid, and along the bottom is the View Control Bar, enabling you to toggle the appearance of other views within the interface. These include the Browser or the Inspector, which can be displayed on the left, alongside a tall, thin Fader view, giving you a channel strip-like control for the currently selected track, as on the Mac version of Logic Pro.

Three buttons in the centre of the View Control Bar set what interface categories to show in the lower part of the display: the Plug-ins area, Mixer, or the Editor, which shows either the Audio, Piano Roll, Drummer Editors, or the Step Sequencer, depending on the type of Region selected »

Apple Logic Pro For iPad
$49 per year

PROS
- Perhaps the best example yet of how to reimagine a mature and feature-rich desktop application for the iPad.
- An intuitive, touch-first user experience ensures the right controls are accessible when required, from setting the tempo to adjusting a plug-in parameter.
- Supports the Apple Pencil for creating and editing automation points.

CONS
- It would be great to see Apple Pencil support extended into more aspects of the user experience.
- Some features required for certain workflows might not be implemented or available right now.

SUMMARY
Logic Pro for iPad blends the professional power of Apple's flagship music production application with the mobile magic of the post-PC era.

Play Surfaces provide an on-screen alternative to MIDI hardware controllers. Here you can see a two-manual keyboard, playing Korg's Module Pro running as an AU Extension.

in the Tracks area. These views can be stacked simultaneously if your iPad's display has sufficient pixel coverage.

A final button to the right of the View Control Bar affords access to the many Play Surfaces. These are on-screen control surfaces that appear below the View Control Bar, making it possible to play and record parts just like you might with a hardware MIDI controller. As well as a piano-style keyboard with optional modulation and pitch-bend wheels, there are also Drum Pad, Fretboard, Chord Strips and Guitar Strips Play Surfaces, all of which are easily configurable, with some being reminiscent of GarageBand's Smart Instruments. A nice touch with the Keyboard and Drum Pad surfaces is their ability to dynamically reveal additional rows of keys or pads as you increase the vertical allowance.

Play Surfaces won't replace the need for hardware controllers — and it perhaps goes without saying that any USB class-compliant MIDI device can be used with Logic Pro — but they can be useful in situations where convenience might not permit the presence of a separate physical device.

Gain Of Function

The idea of tools has been carried over from the Mac version of Logic Pro to the iPad in the guise of so-called functions. Where tools dictate the role of the mouse click, functions inform the purpose of a finger tap.

The Tracks area's menu bar contains several function buttons, specifying what happens when the handles of one or more selected Regions are adjusted. With the default Trim function, you can adjust the start and end points of Regions; but if you want to loop a Region or stretch its contents, the Loop or Stretch functions can be used instead. And if you'd rather make some of these adjustments numerically without using the different functions, you can always access a Region's parameters in the Inspector.

If you select a Region with the Split function, a pair of scissors appears in the upper part of the Region that can be dragged horizontally to set the split point, or vertically downwards to perform the division. If you tap and hold the scissors on a Region, the Tracks area will zoom in horizontally (springing back again when you release your finger), which seems like a great way to perform finer edits. However, if the Snap value is set to, say, a quarter-note resolution, this setting persists even during the brief moment of zoom.

In addition to using the Split function on individual Regions, it's also possible to split many selected Regions simultaneously based on the location of the playhead. Simply select the desired Regions and then tap one of them to bring up the shortcut menu, which offers easy access to several context-specific commands. In this case, tap Split, followed by Split by Playhead. Again, this is pretty much identical in concept to how you might work in Logic Pro on the Mac, except that you'd right-click to open the shortcut menu instead.

Complementing the function buttons are two modifier buttons to assist with selecting multiple objects and copying. As with GarageBand, you can select multiple objects like Regions by tapping and holding the first object, before tapping additional objects with another finger to select those as well. However, you could also do this by enabling the Multiple Select button, which toggles the selection status of objects as you tap them. Furthermore, if you have the Multiple Select button active and you tap and hold an object to select it, you can drag out a selection box to select other, nearby objects. And should you want to create a copy of a Region when you drag it, as opposed to moving it, you can activate the Copy button.

A further behavioural trick offered by the Function and Modifier buttons is the ability to use them as momentary buttons. If you tap and hold a Function or Modifier button, perform the desired action and release the button, the previously enabled button state will be restored. For example, if you want to copy a Region and have the Trim function selected, you could tap and hold the Copy button, drag the appropriate Region, and release the Copy button to re-enable the Trim function button. This is rather neat, although the momentary state only works if you perform an action whilst the button is held down; if no Region had been copied in the previous example before releasing the Copy button, that button would remain enabled when you release your finger.

The ability to use the Modifier buttons momentarily is very much akin to holding

A numeric input dialogue makes it easy to edit parameters, using either the scroll wheel or the numeric pad. Other context-specific controls conveniently appear when necessary, such as the Tap Tempo button when editing the tempo.

System Requirements

Logic Pro For iPad has surprisingly modest requirements, supporting any iPad with at least an A12 Bionic chip running iPadOS 16.4. This means any iPad starting with the third-generation iPad Pro, launched in November 2018 (which featured the more powerful A12X Bionic variant), is compatible, including the third-generation iPad Air and fifth-generation iPad mini, which were both released in March 2019.

To get the most out of Logic Pro For iPad, a fifth- or sixth-generation iPad Pro (powered by an M1 and M2 respectively) is going to provide the best experience; the fifth-generation, M1-based iPad Air affords the best value and is often discounted by retailers like Amazon. If you're considering an M-series iPad Pro, it's worth opting for a device with at least 1TB of storage, since these models feature 16GB of unified memory. The Pro models with less capacity (along with the M1-based iPad Air) include 8GB of unified memory, which limits the number of plug-ins you can use simultaneously in much the same way it would on a Mac.

KIT PLUGINS

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KIT PLUGINS

The Plug-ins area shows each plug-in used in the currently selected track, from left to right, represented as tiles, each showing a custom user interface with the most important parameters visible.

down a modifier key on the keyboard whilst dragging an object with the mouse. And, indeed, if you have a keyboard paired with your iPad (such as Apple's own Magic Keyboard), holding down the Shift or Option keys activates the Multiple Select or Copy Modifier buttons.

Plug It In, Plug It Out

When it comes to shaping the sound of a given track, you can naturally turn to the more traditional Mixer view. However, Logic Pro For iPad offers a new, track-based approach for working with plug-ins, sends and outputs, accessible in the new Plug-ins area in the lower part of the interface. This area is an evolution of Logic Pro's Smart Controls, comprising three views called Track, Sends and Output.

The Track view shows a series of tiles, each representing an insert plug-in, from left to right. This approach is reminiscent of the Device view in Ableton's Live; although rather than each tile showing a full user interface, a few key controls are provided, giving easy access to commonly tweaked parameters. For example, Tape Delay's tile provides four controls: the wet and dry levels, the note duration of the delay, and the amount of feedback. You can always see the full set of controls for a plug-in by tapping its name to open it in Details view. Many of Logic's plug-ins have retained attractive facades, such as ChromaVerb, whilst some of the more generic-looking offerings have been reduced to simplified rows of touch-friendly controls to maximise the available screen real estate.

If an Instrument track is selected, the Plug-ins area's Track view contains additional controls for adding an Instrument plug-in as well as MIDI effects plug-ins. And indeed, Logic Pro For iPad contains the same nine MIDI plug-ins as the Mac version; there's even an implementation of Scripter, which makes me as happy as a littleneck clam.

Adding a plug-in is a simple matter of clicking the Add button, and further options are available by tapping and holding a plug-in's tile. You can quickly access relevant Presets and Patches making use of that plug-in in the Browser, in addition

This screenshot shows how all the user interface areas can be used together as part of a cohesive workflow. Here you can see a Pattern Track in the Tracks area, with the selected Region being edited in the Step Sequencer, along with the Drum Pad Play Surface at the bottom.

to removing or replacing it with another, or inserting a new plug-in before the selected one.

To reorder the plug-in chain, it might be tempting to try tapping and holding a plug-in tile until all the tiles begin to wiggle, but sadly a more mundane approach has been adopted. Clicking the Edit button in the Plug-ins area's menu bar shrinks the tiles horizontally, adjusting their appearance to show just a name and removal button, making it much easier to tap and drag a tile to reposition a plug-in within the chain.

One limitation is that Logic can only address the main output, even if your interface has several. With my RME Babyface Pro (ensuring Logic Pro's 'Run in Background' setting was enabled) I could re-route Logic's output to other outputs via RME's TotalMix FX app, but there's no way to use secondary outputs in Logic Pro for headphone mixes and so on.

Touch & Go & Go, Go

Debuting in Logic Pro For iPad are two new plug-ins with interfaces specifically enhanced for touch: Sample Alchemy and Beat Breaker. Based on Alchemy, Sample Alchemy is a new instrument where you can take a single audio sample and resynthesize it into a playable instrument, using four different generators to transform the source material based on granular, additive or spectral techniques. Beat Breaker is a multi-effect offering a ReCycle-style way to remix drum loops, splitting real-time audio into slices that can be rearranged, repeated, rewound and reworked as desired.

Beat Breaker's Main Editor lets you define patterns that specify how incoming audio is manipulated by the effect, mapping input beats to output beats. Each pattern can contain up to 32 slice markers, which are basically signposts for the start and end points of an output beat.

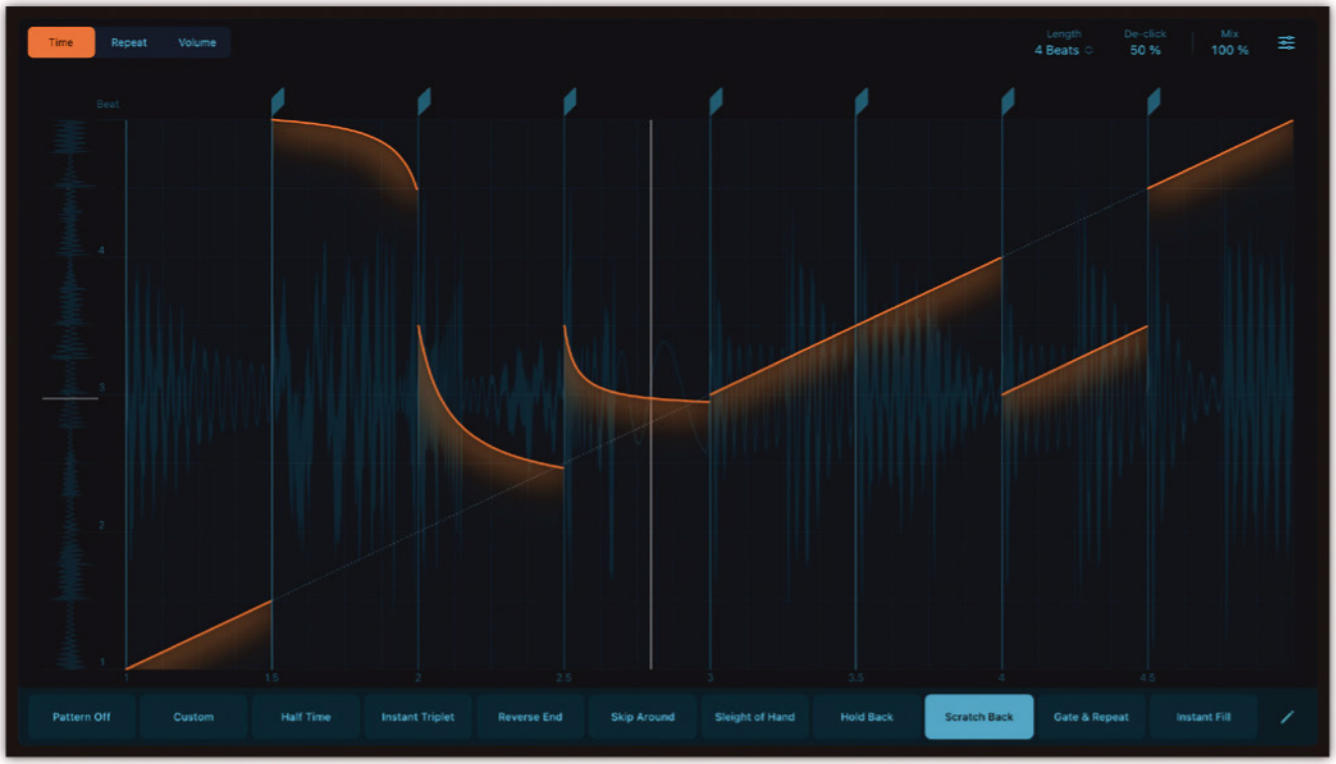
Three different modes are provided to configure how a pattern employs the slices for playback: Time, Repeat and Volume. Time lets you adjust the playback speed of a slice, and map what input beat to trigger as the output beat from a given slice marker. Repeat mode can set how many times a beat is repeated (and thus divided) within its duration, while Volume mode unsurprisingly lets you alter the volume for a given slice, with a slope factor specifying the depth of a fade in or out, based on a curve that sets the rate of the fade. A set of 10 Pattern buttons can be assigned to trigger different patterns stored as presets, along with an 11th button to disable pattern playback altogether, leaving incoming audio to pass through the plug-in unmodified.

Test Spec

iPad Pro (sixth generation), featuring an M2 with 16GB unified memory and 1TB storage, running iPadOS 16.5.1.
Apple Pencil (second generation).
RME Babyface Pro and NI Komplete Kontrol S88.
Logic Pro 10.7.9 for macOS.

The Automating Dance

As you would expect, Logic Pro For iPad makes it possible to automate various mixer and plug-in controls, either by enabling one of the relevant automation modes to record on-screen control movements or by editing automation points manually. And, just like the Mac »



Beat Breaker is a new multi-effect plug-in that can split incoming audio into slices, to be manipulated in a number of different ways.

» version, there are two types of automation available depending on whether you want that data to be associated with a track or a Region.

Track automation is accessible in the Tracks area by tapping the Automation View button on the Tracks area menu bar, whilst Region-based automation editing is handled in the Editors area. This differs in a good way from the Mac version, where both track and Region-based automation are accessible in the Tracks area. It automatically provides a separate and larger area for Region-based automation editing, making it possible to focus on a selected Region immediately without having to adjust the zoom settings of the Tracks area.

With Automation View enabled, the available collection of Function buttons changes to reflect the new editing context, and the Move function is selected by default. You can select an automation point, or the line between two points, by tapping, and then move the selection horizontally and vertically to adjust time and level. Selection of multiple points and lines is handled in much the same way as described earlier for selecting multiple Regions.

You can use the Pencil function to add single points or the Brush function to create multiple automation points as you drag your finger. However, working with automation really comes alive if you have an Apple Pencil. Not only does the Pencil make it possible to edit automation points with far greater precision; it also affords a more natural flow when drawing curves than could ever be achieved with a mouse, trackpad or finger. Working with automation using the Pencil is fantastic, although it's the one area of the user interface where I wish

Audio Unit Extensions

Although Logic Pro supports Audio Units on both the Mac and iPad, enabling third-party plug-ins to be used within your Projects, they're deployed using slightly different implementations. Audio Units conforming to the v2 specification have been the most prevalent type on macOS, where a component (or bundle) is installed into the Library/Audio/Plug-Ins/Components folder. However, macOS 10.11 and iOS 9 (before iPadOS became the dedicated version of iOS for iPads) introduced Audio Units v3 (typically abbreviated as AUv3), allowing developers to deliver custom instruments and effects via an App Extension.

The advantage of an App Extension is that, by default, such Audio Units will run in a separate process to the host app, meaning that a crash won't necessarily bring down the host app. And because App Extensions are contained within an app, they can be distributed through the App Store.

Understanding this distinction is important because while macOS applications can host both v2 and v3 Audio Units, only AUv3 Extensions are supported on iPadOS (and iOS). Moreover, just because an Audio Unit runs natively with Apple Silicon on a Mac doesn't mean it will be available on iPadOS. Only Audio Units that are both compatible with iPadOS and distributable via the App Store can be utilised by Logic Audio for iPad.

For mostly technical reasons, Audio Unit Extensions have yet to be widely adopted on the Mac, meaning the majority of Audio Units are still of the v2 variety. Nevertheless, the App Store contains many AUv3 Extensions compatible with iPadOS from developers such as FabFilter and Baby Audio, since existing host apps like GarageBand and Steinberg's Cubasis have offered support for some time.

Unfortunately, though, this bifurcation in the Audio Units ecosystem currently makes life more difficult. For example, if you're looking to embrace Logic Pro's Roundtrip capabilities, you may need to repurchase existing plug-ins as AUv3 Extensions from the App Store to use them on both platforms. This is going to be a real pain in the short term for many users, but it's hard to point the finger at Apple or third-party developers, given how the Mac and iPadOS platforms have naturally developed over the years.

The good news, though, is that this situation should improve over time. Common developer frameworks like JUCE offer support for AU Extensions, and Steinberg's official VST3 SDK enables VST3 plug-ins written with Steinberg's tools to be more easily ported as AU Extensions, so third parties should find the task of supporting the iPad less daunting.

Apple had gone even further. For instance, the second-generation Pencil supports a double-tap feature to change tools; the ability to switch between the Brush and Eraser function by double-tapping the side of the Pencil really would be the cherry on the icing on the cake.

Taking A Roundtrip

Perhaps the feature on which Apple are taking the biggest gamble with Logic Pro For iPad is Roundtrip, which gives you the ability to work on the same projects using either the Mac or iPad versions. And I use the word 'gamble' because there are so many factors both within and outside of Apple's control to ensure this works correctly and as intended.

In a perfect world, Logic Pro For iPad can open projects from the Mac version that have been saved as Packages (which contain all necessary Project data, such as audio files and other related content, within a single bundle) as opposed to a Folder. Conversely, Logic Pro for macOS can open projects from the iPad version, which are only saved as Packages. And if you're storing these projects using iCloud, this can work seamlessly from a file management perspective.

However, not every Project can be opened with Logic Pro For iPad, even when saved correctly as a Package. For example, since some features present in the Mac version aren't currently supported on the iPad, such as Spatial Audio or surround in general (see the 'Logical Omissions' box), Projects that use these features can't be opened by Logic Pro For iPad. (Apple have detailed a list of such incompatibilities at https://support.apple.com/en-us/HT213666.)

When Roundtrip works flawlessly, it's incredible; when there are issues, it can be incredibly frustrating. As mentioned already, many of the factors contributing to a Roundtrip disruption are outside of Apple's control, and this is particularly true when Audio Unit Extensions are involved (see box). Therefore, how useful Roundtrip will be to you very much depends on the type of Projects you wish to take from your Mac and open on an iPad. If you plan on mostly starting Projects on your iPad whilst you're out and about, before bringing them into Logic when back in the studio, the compatibility will be virtually seamless, especially if you're not using any third-party plug-ins.

Not Your Daddy's Logic Pro

Logic Pro For iPad is a *tour de force*, demonstrating how a professional music production app should function on a device like the iPad. To say I was blown away by the user interface and its interaction design would be a bit like saying I thought the pontiff was a tiny bit Catholic. And Apple have achieved this without sacrificing sophistication, with Logic Pro For iPad offering almost everything you could want: from Live Loops to Track Stacks, Global tracks for tempo changes, Markers, and a full complement of MIDI and audio effects and instruments along with the associated content libraries.

One question I've encountered in conversations is whether Logic Pro For iPad could be used instead of Logic Pro on the Mac. But I'm convinced this is very much the wrong question to ask. I don't think you would use Logic Pro on the iPad instead of using Logic Pro on the Mac; you'd use Logic Pro on the iPad where it makes sense to be working with that app on that device, and I'm quite sure Apple's intention isn't for one to replace the other. Indeed, Roundtrip with iCloud support is an obvious example of how the two platforms can be used in concert with each other. And the fact Logic Pro can now be used on a device with a form factor offering even greater mobility and affordability than a MacBook just opens music production to more people and usage situations.

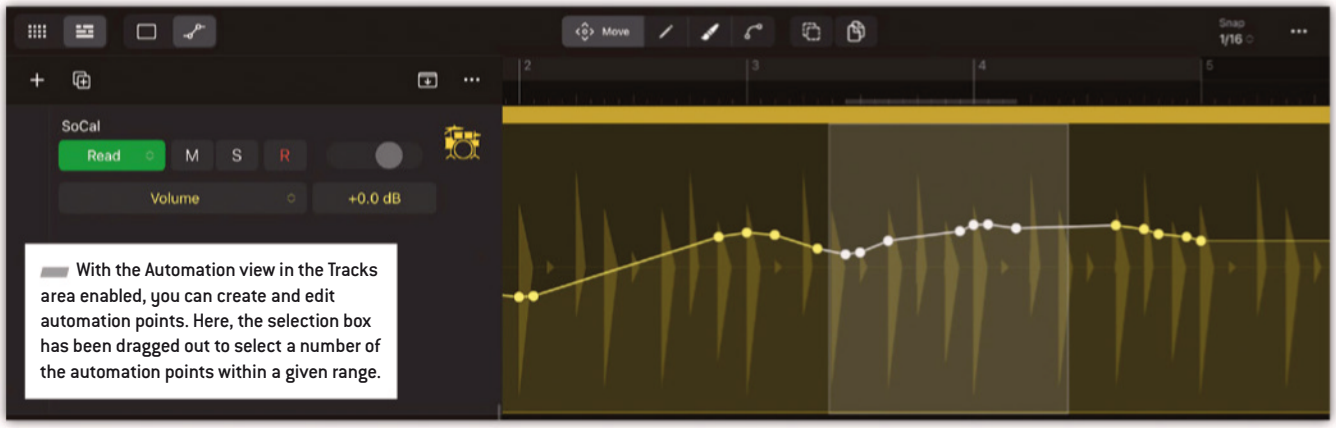
Some will inevitably complain the moment the word 'subscription' is mentioned, but in this case I really think it would be churlish to protest about the price. When you consider the features and content on offer, and how easy Apple have made subscription management in iPadOS's Settings app, subscribing and paying for Logic Pro is likely to be far less frustrating and expensive than signing up for a newspaper.

$ $4.99 per month or $49 per year
W www.apple.com

Logical Omissions

Not every feature from the Mac version of Logic Pro is currently accessible or implemented in the iPad version. This is hardly surprising in the first release of the app; in any case, some features simply might not make sense for the iPad version. For example, there are sophisticated notation apps already available for the iPad, so it's hard to see why it would have made sense for Apple to make scoring a priority. The lack of support for Spatial Audio (and surround in general) is perhaps a little more surprising, given the emphasis Apple have placed on these technologies generally, but I don't think there are many potential users who will see this as a deal-breaker.

While Logic Pro For iPad benefits from a full complement of plug-ins, some of the effects and instruments are primarily included for preserving Roundtrip compatibility, especially those requiring more complicated user interfaces for editing. This affects Sampler, Alchemy, and — to a lesser extent — Scripter, which each offer simplified interfaces intended for playing back existing content. Ultimately, though, I feel Apple have made the right choices in deciding what features to include in this first release of Logic Pro For iPad, balancing its functionality between the necessary and the nice to have.



With the Automation view in the Tracks area enabled, you can create and edit automation points. Here, the selection box has been dragged out to select a number of the automation points within a given range.

Austrian Audio MiCreator

Modular USB Microphone System

Austrian Audio's MiCreator is more than just a USB mic. It's an entire family of mics!

SAM INGLIS

Sooner or later, it seems, almost all professional microphone manufacturers succumb to the temptation to produce a USB microphone. Margins are slim and the competition cut-throat, but the potential market is vast compared with recording studios, broadcast or live sound. Understandably, the resulting mic is often derived from an existing product. Audio-Technica's AT2020USB+, for example, is a version of their very popular AT2020 outfitted with a USB port, a built-in preamp and a headphone socket.

Austrian Audio are the latest manufacturers to climb aboard the good ship USB, but they've taken a very different approach. MiCreator is not a single product but a modular system, and it bears no resemblance either visually or functionally to any existing Austrian Audio mics. (Whereas Austrian Audio's high-end mics use capsules manufactured at their Vienna HQ, the more affordable MiCreator range is made in China and uses third-party capsules.)

Forward & Back

At launch, there are three principal MiCreator products, and the core of the system is the MiCreator Studio. Cosmetically, it's a striking hybrid of ultra-modern and ultra-vintage; the headbasket design hints at the carbon button mics of the '30s, but the overall vibe is unmistakably 21st Century. In practice, the ring-shaped capsule mount offers effective built-in shockmounting, and allows the capsule to be tilted through 30 degrees or so either side of vertical.

The body of the MiCreator Studio has roughly the proportions of the monolith from *2001: A Space Odyssey*, and is reassuringly weighty. It's designed to stand upright on a table or desktop, but a 3/8-inch socket in the base allows it to be attached directly to a mic stand. A 3/8- to 5/8-inch adaptor is supplied, but there's no easy way to attach it to a camera tripod, which might be a more valuable option for some of its target market. If the all-black look is too solemn for you, it's possible to switch out the front and rear faceplates for more colourful alternatives. Red plates are supplied as part of the System Set (see box), and it will also be possible to buy plain black and white ones that can be customised.

The faceplates are made of rigid plastic and attach magnetically. Holes and slots accommodate the MiCreator Studio's various sockets and controls, of which there are more than you might expect. On what you soon discover is the rear panel, you'll find a Type-C USB port, along with mini-jack sockets labelled 'out' and 'in/out'. The former is, as you'd expect, a headphone socket, while the latter is the portal to the wider MiCreator ecosystem. It can serve as a second headphone port, but its primary role is for

Austrian Audio MiCreator

From $199

PROS

- Much more versatile than most USB mics.
- Easy to use.
- Supports two pairs of headphones.
- Satellite can be used standalone.

CONS

- The low cost of the Y-Lav is somewhat reflected in its design.

SUMMARY

Austrian Audio have thought hard about what content creators might want from a USB mic, and they've come up with an innovative modular system that ticks a lot of boxes.

connecting a MiCreator Satellite, MiCreator Y-Lav or guitar. (No additional outlay is needed in order to plug a guitar in, as a suitable cable is supplied.)

Turn to the front of the MiCreator Studio and you'll see a three-position sensitivity switch offering high gain, low gain and mute settings, plus a rotary control with a push action. In normal service, this governs the volume of the headphone output, which is reported visually using a strip of seven green LEDs. Give the rotary a confident push and the LEDs turn red, indicating that it is now controlling 'balance'. Specifically, it adjusts the relative levels of the MiCreator Studio's built-in mic and whatever is plugged into the rear 'in/out' socket. Playback level from your DAW is fixed, so if you don't have a second MiCreator source plugged in, you can use this mode to adjust the balance of input signal against playback. With a second source, you can only have more of one input at the expense of less of the other! Both inputs are always panned centrally, which probably represents a reasonable compromise in terms of ease of use.

The MiCreator Studio appears to your computer, phone or tablet as a class-compliant USB device offering stereo input and output. There's no control panel, and no drivers are needed on macOS or iOS. Sample rates of 44.1 and 48 kHz are supported.

The MiCreator Family

The MiCreator Satellite is the MiCreator Studio's little brother, sporting the same ring-mounted capsule assembly atop a sawn-off body. Again, there's a 3/8-inch standmount socket in the base, and both front and rear faceplates can be swapped out and customised, but there's no volume control, just two mini-jacks labelled 'link' and 'out'. The former is used to connect the Satellite to the Studio, using a cable with TRRS plugs at either end, while the 'out' socket is once again intended for headphones. Once the two are connected, this duplicates what comes out of the headphone socket on the Studio, and is governed by its volume control. The gain of the Satellite or Y-Lav is governed by a two-position switch on the Studio's rear »

The MiCreator Studio is the hub of the system.

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Howdy Neighbors!

The MiCreator Satellite is designed to connect to the Studio, but can also be used standalone with devices that offer plug-in power.

» panel, which simply offers high and low options — once connected, the secondary source can't be muted. The Satellite can also be used outside the MiCreator ecosystem, as a standalone device with cameras, phones, laptops and other devices that can supply 'plug-in power'. A short splitter cable is supplied for use with cameras.

The Y-Lav is the simplest of all the MiCreator devices. As the name suggests, it's an omnidirectional lavalier mic, which attaches to clothing or other suitable targets using a sprung clip. A thin 2m cable is hardwired to the mic at one end and, at the other, to one side of a moulded plastic splitter. The other half of the split feeds a mini-jack headphone socket, while a TRRS mini-jack plug connects to the Studio's 'in/out' socket. A removable foam windshield is included.

The MiCreator Y-Lav is a lavalier mic with its own headphone socket.

Quality-wise, the Y-Lav was probably the element of the MiCreator system that impressed me the least. I suppose making the cable non-detachable eliminates one potential source of problems, but it means that if it fails, you'll have to throw the entire thing away. And whilst the build quality is neat enough, it's not as though Austrian Audio have used armoured cable or super-duper strain relief. You'll need to take good care of it if you want it to have a reasonable lifespan. The mic doesn't rotate on its mount, which can make finding the optimum position harder, and although it works perfectly well, it's neither the quietest nor the most sonically forgiving lav mic I've tried. Given the price, it would be unfair to expect the same quality you'd get from a high-end DPA or Sony model, but if you're a regular user of such things, it'd be worth considering a more pro-oriented option.

Star Links

The Satellite, meanwhile, is sonically identical to the Studio, which means that it's a cardioid small-diaphragm electret capacitor mic. Some low-end roll-off is built in to compensate for the proximity effect, giving it a flat bass response in close-up use. Along with the multi-layered basket and integral shockmount, this makes the MiCreators relatively immune to thumps and plosives. Self-noise is specified at 22dBA, which is higher than in a typical studio mic but not so high as to be problematic in close-up use, and the two fixed sensitivity levels seem well chosen. I found that the higher of the two was about right for speech, quiet singing, lightly strummed acoustic guitars and so on, whilst the low sensitivity setting will accept SPLs of up to 130dB, so should cope adequately with small guitar amps.

Tonally, the Studio and Satellite are quite forward when used on-axis, with a reasonably lively presence boost in the upper midrange. They aren't as smooth as Austrian Audio's own CC8 pencil mic, but given the price differential, that's hardly surprising! Assuming the idea is to deliver a clear, 'mix ready' sound for streaming and voiceover work, I'd say the voicing is pretty well judged.

Of course, if you just want to record or stream a single voice, there are a bazillion other USB mics out there, and quite a few of them are even more affordable than the MiCreator Studio. What makes this product unique is the versatility that the 'in/out' socket gives you. Right out of the box, you have the option to connect a second pair of headphones, or use the supplied instrument cable to record DI guitar along with your voice. Add one of the other MiCreator products and you open up a wide range of other possibilities. You could, for example, record the same voiceover artist simultaneously with both the MiCreator Studio and a Y-Lav, and decide after the fact which one has delivered the best sound. Or you could bring in a second speaker and give them the Satellite or Y-Lav, plus their own set of headphones. My favourite use case in a musical context, though, is stereo recording. The MiCreator System Set comes with a specially designed stereo bar called the Minibar that allows a Studio and a Satellite to be used as a spaced pair, and this arrangement gave me an excellent acoustic guitar recording with no fuss at all, nor any fiddling around with gain controls.

Austrian Audio haven't simply produced a me-too USB mic because they feel they ought to have one in their range. Instead, they've applied some fresh thinking to the concept, and they've come up with an innovative modular system that can support a surprising number of applications. It'll be interesting to see what else they have in store for future MiCreator products.

MiCreator Pricing

The three elements of the MiCreator system are available separately, or as part of the MiCreator System Set, which includes all three plus a Minibar stereo bar, cables, adaptors and the alternative red faceplates. Buying a MiCreator Studio or System Set also gives you a licence for Steinberg Cubase LE or WaveLab Cast. Prices include VAT.

- MiCreator Studio: $199
- MiCreator Satellite: $99
- MiCreator Y-Lav: $49
- MiCreator System Set: $299

Accessories bundled with the MiCreator Studio include a special cable for DI'ing a guitar.

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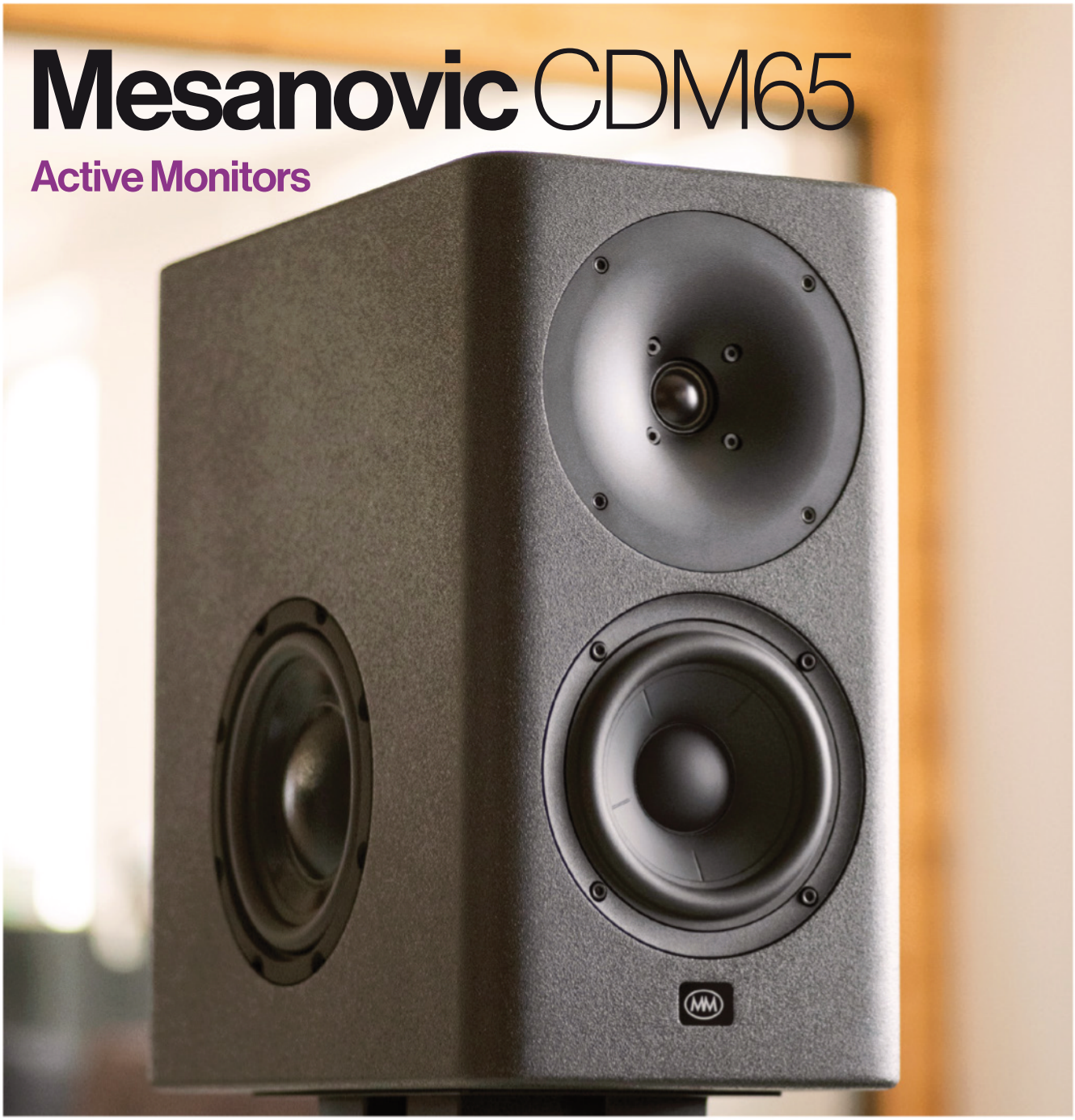
Mesanovic CDM65

Active Monitors



Mesanovic's three-way speaker offers outstanding performance, wireless connectivity — and a cardioid bass response.

PHIL WARD

I reviewed boutique US microphone manufacturers Mesanovic's first foray into monitor manufacture, the RTM10, back in *SOS* July 2021. In the spotlight this month is the company's second monitoring product, the CDM65. And, ignoring the fact that all moving-coil speakers are conceptually similar, the CDM65 and RTM10 are like chalk and cliché. I say this because whereas the RTM10 has just an analogue input, the CDM65 adds the WiSA HT wireless audio protocol and all the streaming options that this technology makes possible. And where the RTM10 incorporates an in-house developed and manufactured ribbon tweeter, the CDM65 employs a considerably more conventional OEM device. There's more differentiation, too, because where the RTM10 is an entirely conventional two-way direct-radiating loudspeaker with acoustic dispersion defined by its drivers and enclosure dimensions, the three-way CDM65 features Kii Three-style active cardioid radiation, enabled by its side-mounted drivers functioning as both low-frequency radiators and low-midrange dispersion modifiers. So, not only is the CDM65 not much like its Mesanovic sibling, it's a pretty unusual monitor full stop.

Sizing Up

On arrival, the CDM65 turned out to be more compact than I had anticipated. This is a good thing, and means not only that the CDM65 fits easily into a nearfield installation

(although bear in mind that side-firing drivers need a little space to breathe), but also that it's potentially suited to immersive multi-channel installations — tapped inserts on its back panel enable the attachment of VESA standard mounting brackets. Compact does not mean lightweight, though, and at 14.5kg, the CDM65 is still a significant lift on to the monitor shelf. The CDM65 cabinet itself is a slim, rounded-corner, wood-based design finished in a textured dark grey paint. It's the kind of utilitarian paint finish that is both relatively inexpensive to apply and good at hiding material flaws. But, given that the CDM65's wireless functionality seems intended to fit it for consumer as well as professional markets, I wonder if it looks classy enough. Time, and customers, I guess, will tell.

The CDM65's active electronics comprise Class-D amplification rated at 150 Watts each for the midrange driver and tweeter, and 300 Watts for the twin bass drivers. Being a wireless-capable monitor, the signal flow is digital throughout and capable of up to 96kHz/24-bit sampling rate and word length. Around the back is a configuration and connection panel that offers a power switch, a balanced XLR analogue input, a wireless pairing button, an input sensitivity switch (+6dB, 0dB, -6dB) and a USB socket for firmware updates. To switch the CDM65 between its default cardioid-bass mode and 'standard' mode, you need to load alternative firmware from a USB pen drive. A schematic shows a 7.1 speaker system with LED indication that's used to specify the role of the specific CDM65 in a wireless multi-channel system. I'll describe the WiSA HT wireless and multi-channel capabilities more fully in the 'Without Wires' box.

Driving Force

In terms of its electro-acoustic architecture, the CDM65 is a three-way system with four drivers: a waveguide-loaded 25mm tweeter, a 130mm midrange driver and two 165mm side-mounted bass drivers. Rather than being built in-house at Mesanovic, as is the tweeter used in the RTM10, all of the CDM65 drivers are OEM devices, although the tweeter employs a Mesanovic-designed and manufactured waveguide. Said waveguide is CNC machined from solid aluminium and carries a high-quality anodised finish. The aluminium-diaphragm midrange driver appears to be the same as, or at least very closely related to, the one used in the RTM10. It incorporates specific features within its motor system (the voice coil, magnet and associated metalwork) designed to counter some of the distortion-producing mechanisms inherent to moving-coil drivers, and over the decades I've come to realise just how significant these kinds of features are. This bodes well. More generally, the CDM65 midrange driver is one I already know to be of very high performance, both objectively and subjectively.

The side-mounted bass drivers are Kevlar-reinforced paper-diaphragm devices, engineered specifically for their low-frequency role, with four-layer voice-coil windings and relatively generous (±7mm) maximum linear displacement capability. The drivers share a closed-box enclosure of around 14 litres internal volume. Bearing in mind the impressive low-frequency specifications of the CDM65 (-3dB at 36Hz combined) with a maximum of 103dB SPL at 1m, the diaphragm area of the two bass drivers is relatively modest so they will potentially find themselves working pretty hard.

Hot Fuzz

Now seems a good time to describe and explain how the drivers are employed to create the CDM65's cardioid radiation, and as part of that, we'll look at some FuzzMeasure acoustic data. Being a three-way system, the CDM65 has two crossover frequencies, where complementary low-pass and high-pass filters are used to integrate the radiation of each driver. The low-frequency to mid-frequency crossover of the CDM65 is at 150Hz, and its mid-frequency to high-frequency crossover is at 2.3kHz. »

Mesanovic CDM65
$5298

PROS
- Wide bandwidth.
- Active cardioid.
- Fabulous imaging and low coloration.
- Neutral tonal balance.
- Great bass.

CONS
- None.

SUMMARY
There's no difficult second album for Mesanovic. Their second monitor is a brilliant example of the electro-acoustic arts, with a neatly realised and effective cardioid trick for good measure.

As well as featuring an analogue XLR input, the CDM65 can be used wirelessly as part of a WiSA network in configurations up to 7.1 surround.

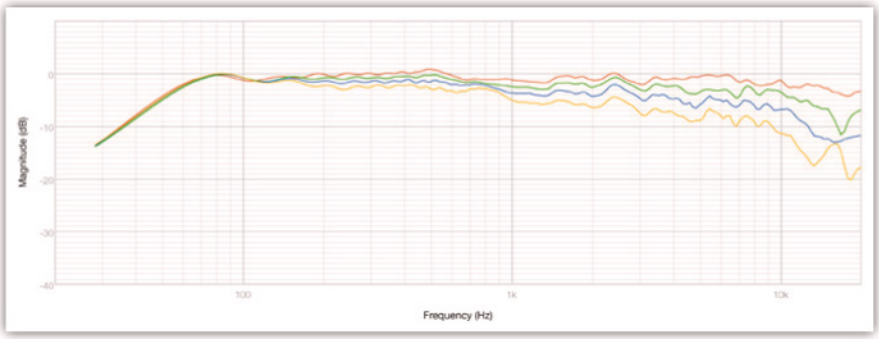


Diagram 1: The CDM65's frequency response in cardioid mode, measured on axis (red trace), and 30, 45 and 60 degrees horizontally off axis (green, blue and orange, respectively).

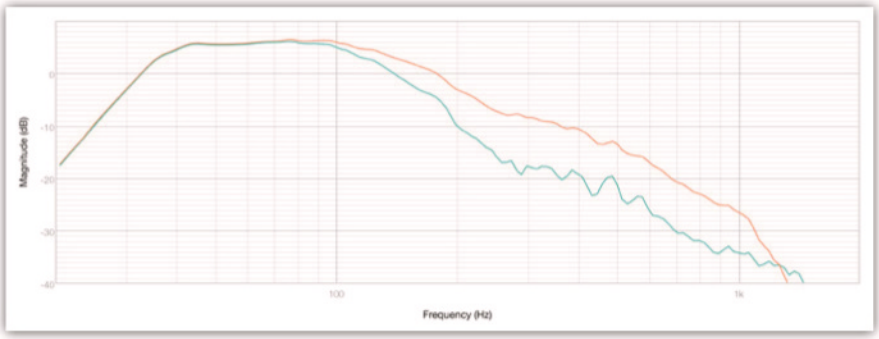


Diagram 2: The CDM65's bass driver output, measured at 0.5cm and in both standard and cardioid modes (green and red traces, respectively).

Now, if the CDM65 were a conventional monitor with dispersion primarily defined by its physical dimensions, that dispersion would be substantially omnidirectional up to around 500Hz, where the wavelength begins to become comparable to the cabinet dimensions.

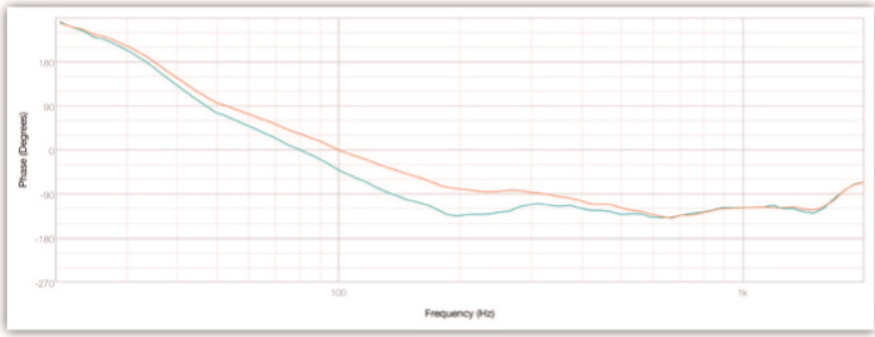
However... Diagram 1 illustrates the CDM65 cardioid mode dispersion through frequency response curves measured on axis and at 30, 45 and 60 degrees off axis horizontally, and the CDM65 clearly begins to become directional at the advertised 150Hz. That 150Hz also marks the bass-to-midrange crossover frequency is no coincidence because, in cardioid mode, rather than quickly being attenuated with a steep low-pass filter, the bass drivers are rolled off slowly so that their output overlaps in the 150Hz-1kHz band with that of the midrange driver. The overlap results in interference between the bass and the midrange drivers, and the result of that, modulated by a degree of time-domain massaging of the bass driver output above 150Hz, is the selective modification of the system's inherent directivity right down to the 150Hz point.

To illustrate this further, FuzzMeasure Diagram 2 shows the 20Hz-2kHz output of a CDM65 bass driver, captured by placing a measuring microphone around 0.5cm from the diaphragm, with the speaker in cardioid (red curve) and standard (green curve) modes. The difference in the bass driver roll-off is clearly apparent. Diagram 3 also records the close-mic bass driver output, but this time shows phase rather than amplitude response. Again, a difference is apparent between the cardioid and standard modes, and it is this modified phase relationship between the outputs of the midrange and bass drivers that results in a cardioid radiation.

I've included one more FuzzMeasure diagram to illustrate the two modes. Diagram 4 shows the in-room frequency response of one CDM65 at the listening position in my studio in both standard and cardioid modes. There's a measurable difference between the two modes, with the most subjectively significant divergence I think being that at around 300Hz.

The final FuzzMeasure curve I've included, Diagram 5, isn't particularly relevant to the CDM65's cardioid characteristics but simply records the monitor's axial frequency response linearity

Diagram 3: The bass driver's phase response, again in both standard and cardioid modes (green and red, respectively).



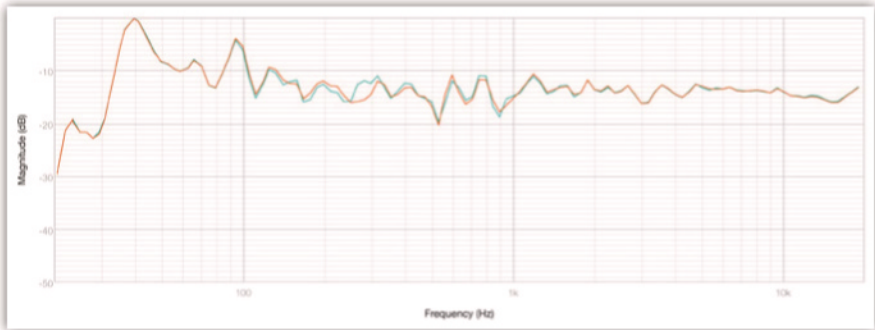


Diagram 4: The in-room response of the CDM65, in standard (green) and cardioid (red) modes.

ALTERNATIVES

The CDM65 is one of those rare monitors for which there isn't much competition. The only other pro monitor that comes close in terms of offering active cardioid electro-acoustics and integration with audio streaming is the **Kii Three** when combined with the Kii Control, but that system costs very much more than the CDM65. A few conventional monitors at around the same price of the the CDM65 that would all be worth a listen are the **PSI A21M**, the **Genelec 8341A**, the **Barefoot Footprint 01** and the **Amphion Two15** with its matching **Amp700**.

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» and harmonic distortion performance. Both are very impressive. The frequency response is predominantly flat within around ±1dB, and the distortion products, measured at 90dB output level at 1m are, below 100Hz, typically 50dB down (0.3%) in the case of the second harmonic and 60dB down (0.1%) in the case of third harmonic.

In Use

And so to listening. I set up the CDM65s on my monitor shelf in prime position, temporarily demoting my current Neumann KH150 reference and Dynaudio BM5 MkIII Atmos monitors to the floor. As usual, I played a bunch of familiar material, and very much enjoyed the experience, because the CDM65 is an extremely fine monitor — up there with the best I've heard in recent years. It's the kind of monitor that makes this review job so enjoyable (because when a monitor is this good, I quite quickly default to listening just for pleasure), but at the same time makes it somewhat difficult to know what to write because flaws, or even just quirks, are often the reviewer's raw material and the CDM65 is very short of either. I'm left trying to describe virtues, and that's more difficult to do without venturing into hi-fi review territory and turning the descriptive hyperbole up to 11. So I'm going to keep it relatively short.

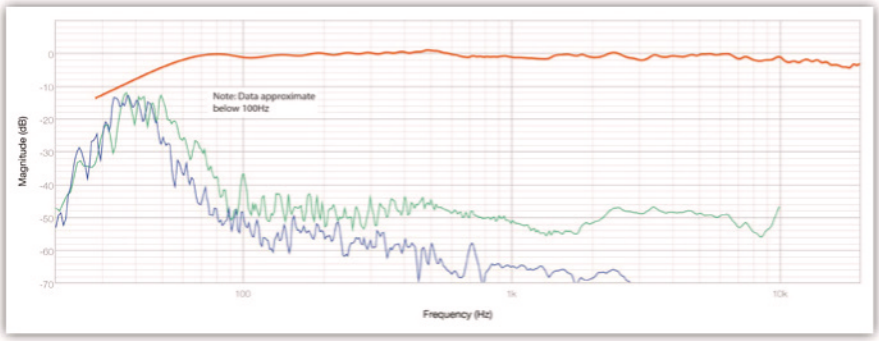

Diagram 5: The on-axis frequency response (red trace), overlaid with second- and third-harmonic distortion levels (green and blue, respectively).

Starting at the low end, I was particularly struck by how well the CDM65 reproduces pitch. Low-frequency pitch isn't easy for a speaker because the frequencies that represent different notes are relatively close together and, by the very nature of the the electro-acoustics involved, adjacent to a major system resonance: that of the bass drivers in the enclosure (to which a reflex resonance might be added). But the CDM65 reproduces low-frequency pitch with a notable level of explicit accuracy. It makes for a really secure foundation for a piece of music. Low-frequency timing and dynamic accuracy is also very good, but these things do tend to go hand in hand with good pitch reproduction.

Moving up the audio band, the midrange driver is uncoloured and displays a natural, even-handed quality on vocals and acoustic instruments that suggests that it'll be trustworthy in translation terms on everything else too. It also has a highly revealing nature that very effectively lays all the details bare. And finally, at high frequencies, the CDM65 tweeter is the epitome of clarity and detail without emphasis. But if there's anything that sums up just how good a monitor the CDM65 is, it's the way in which all the different elements are well integrated, yet get out of the listener's way to reveal exactly what's going on in the mix.

As far as the subjective difference between cardioid and standard mode is concerned, even though the relatively long delay involved in switching from one mode to the other makes it hard to A/B them directly, the contrast is marked. In tonal terms, standard mode sounds a little warmer in the voice band than cardioid mode, and in imaging terms, cardioid mode is noticeably more tightly focused, with a more explicit portrayal of stereo depth and perspective — to the point where the imaging is truly striking. Cardioid mode sounds generally more analytical and better suited to a mix role, and after a few mode swaps, became my preference. But even in standard mode, the CDM65 is still exceptional.

The CDM65 is a hugely capable monitor that offers very wide bandwidth, a revealing and uncoloured midrange, a high end that's detailed and delicate without drawing undue attention, and brilliant stereo imaging. It's is a fine example of the speaker designer's art and it makes for supremely effective monitoring. And if it floats your boat, and suits the way you work, it can play the consumer audio multi-input, multi-channel wireless streaming role too.

$ $5298 per pair.
W www.mesanovicmicrophones.com

Without Wires

The WiSA acronym stands for Wireless Speaker and Audio Association and, along with its conventional balanced analogue input, the CDM65 employs the WiSA protocol to enable wireless operation. The WiSA association is a group of large and primarily consumer-focused electronics brands and their technology is becoming increasingly visible in consumer multi-channel audio systems. Rather than employing a local WiFi network or Bluetooth to transmit and distribute audio to wireless playback devices, the WiSA HT hub hardware creates a separate dedicated wireless mesh network, and that confers the system with some notable advantages. Firstly, because WiSA HT doesn't rely on an existing WiFi network, it doesn't suffer from network congestion. Secondly, WiSA HT is able to handle sample rates and word lengths up to 24-bit/96kHz, and lastly, WiSA HT isn't limited to stereo; it can transmit audio to playback system formats up to 7.1. A 7.1-channel format unfortunately doesn't cover Dolby Atmos mix work, where the minimum speaker-based monitoring system is generally accepted to be 7.1.2 or 5.1.4. Of course, there's nothing to stop anybody building an Atmos monitoring system based on seven wirelessly connected CDM65s in, say, a 5.1.2 format, but it will be an outlier in terms of Atmos mixing. Having said all that, I think it's probably the case that Mesanovic see the CDM65's wireless abilities as more relevant to consumer applications than professional monitoring. I don't imagine there'll be many mix engineers drawn to using CDM65s in a wireless multi-channel mix role.

Mesanovic offer a WiSA hub for the CDM65, based on a device manufactured by Platin Audio (www.platinaudio.us), which offers multiple analogue and digital audio inputs along with compatibility with Apple AirPlay, Spotify Connect and ChromeCast. I was able to try a Platin hub for the very last couple of days that I had the CDM65 review samples. Once I'd worked through the WiSA configuration and pairing process using the Mesanovic app, the hub worked flawlessly and I soon had both a turntable and Spotify from a phone happily playing though the CDM65s. It would also of course be feasible to send the output of a recording interface to the WiSA hub and use the CDM65s as wireless monitors. In that mode of use, the EQ and room optimisation functions (the latter based on smartphone mic data capture) that the Mesanovic app offers, but which aren't available on the CDM65 analogue inputs, might add an attractive further element of monitor configuration.

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For this month's exclusive *SOS* competition, we've teamed up with Vienna-based microphone innovators LEWITT Audio to offer not just one, but three fantastic studio tools.

First up is the LCT 440 PURE. Despite being one of LEWITT's more affordable offerings, this large-diaphragm cardioid capacitor microphone still bears all the hallmarks of the company's innovative approach to microphone design, from its stylish looks to its bespoke shockmount and magnetically attaching pop filter. It even employs the same capsule as LEWITT's high-end designs.

Next we have the LCT 640 TS, a multi-pattern, dual-diaphragm mic with a difference. Unlike most multi-pattern mics, the LCT 640 TS gives you access to the outputs from both diaphragms independently — meaning you can record sound from both the front and rear of the mic at the same time, and even change the polar pattern after recording. Picked up too much room sound recording acoustic guitar in omnidirectional mode? No problem: just fire up the free Polarizer plug-in in your DAW and change it to cardioid or figure-8!

Finally, the CONNECT 6 takes LEWITT's novel design approach and applies it to the world of audio interfaces. Its two premium preamps can dish out a massive 72dB of gain if required (enough for even the quietest dynamic mic), while flexible routing and DSP allows for any music or podcast recording workflow, and even includes built-in EQ, compression and Maximiser processing.

Uniquely, the CONNECT 6 features no fewer than three USB ports round the back. One is for connection to a computer, as you'd expect, and the interface can be bus powered so this may be the only connection you need. And for use with mobile devices, there's a separate USB socket to provide power. Speaking of mobile devices, the third USB port is dedicated to tablets and phones, and you can even use such devices at the same time as connecting to your computer. So, whether you want to pipe your favourite iOS synth into your DAW, or record a phone call as part of a podcast, the CONNECT 6 has you covered.

To be in with a chance of winning one of these three fantastic prizes, simply follow the URL shown, and answer all the questions there, by Friday 1st December 2023. Good luck!

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Clavia Nord Stage 4 88

Stage Keyboard

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GORDON REID

When I started out with this musician lark, every type of keyboard sound I wanted came from a dedicated instrument. If I needed a piano sound, I had to find an electric piano of some sort. If I wanted ensemble sounds, I had to blag a Mellotron or one of those newfangled string ensemble thingies. And if I wanted organ sounds, I had to have an organ. But in 1975, a new class of keyboard appeared. This was the multi-keyboard, so called because it was capable of producing the sounds of multiple classes of instrument, sometimes one at a time but, more often than not, allowing players to access two or more sounds simultaneously using early implementations of what we now call splits and layers.

At the start, these instruments tended to hail from Italy and were — not to put too fine a point upon it — rather cheap and cheerful. But within a year or two, Japanese manufacturers were getting in on the act, with instruments ranging from some relatively small and affordable Korgs at one end of the scale to the monstrously large, heavy and expensive Yamaha SK50D at the other. By providing some combination

Clavia Nord Stage 4 88
$5699

PROS
- The new architecture is a big step forward from previous incarnations.
- There are significant advances in the piano, organ and synth engines.
- Genuinely multitimbral effects — yippee!
- The sound quality is excellent.
- Playing one makes you feel good.
- It remains the pre-eminent stage keyboard for those who don't require a fully-fledged workstation.

CONS
- There remain some unexpected shortcomings.
- The previous 76-note version has lost three keys.
- The Nord Triple Pedal is still an expensive extra.
- It's more expensive than ever.

SUMMARY
If someone offered to buy me a single multi-keyboard to be used primarily for piano, the Nord Stage 4 88 is the one that I would choose. If organ swipes and blistering synth solos were more important, the Nord Stage 4 Compact would also be high on my list. If my own money were involved, the price might force me to cast my net more widely.

of analogue pianos, half-decent organs, fully-decent strings, and brass (which later morphed into paraphonic synthesis) they were a boon for players whose budgets stretched to a tiny fraction of the cost of a Rhodes, Hammond, Mellotron or polysynth (let alone all four). However, the era of the multi-keyboards was short-lived and their fate was sealed in 1981 when affordable polyphonic synthesizers began to appear. Except that it wasn't. If you had told me when I bought my first Juno-60 that the spiritual successor to the Crumar Composer or Multivox MX-3000 would be one of the most widely used and recognised keyboards of the 21st Century, I might have questioned your sanity. Nevertheless, a squizz at the stage of any major gig or festival will demonstrate that the various generations of Nord Stage (which comprise pianos, organs, polyphonic synthesis and effects) are almost ubiquitous within the rigs of those who can afford them. But why? Workstations such as the Korg Kronos and Kurzweil K2700 can do just about everything that a Nord Stage can, plus a great deal more, and cost you less while doing it. So what's the attraction of the big red Swedish beasties?

I think that it boils down to four things: the build quality, the sound quality, the relative simplicity of use and — in a somewhat circular fashion — the aforementioned ubiquity. Turning to the first of these, I'm not aware that anyone has ever seriously questioned the build quality of a Nord Stage. Sure, you have the right to dislike it in matters of personal taste but, in all fairness, you can't claim that it looks or feels anything other than high quality. The same rationale applies to the sound quality. You are under no obligation to like the sounds it produces, but I doubt that many would agree if you were to suggest that it's fundamentally naff. The question of simplicity is itself simple to understand. By limiting the Nord Stage to a handful of mainstream sound engines, Clavia removed much of the fear that sometimes accompanies powerful workstations. The final argument is perhaps the least obvious but, if your live rig is based upon a Nord Stage, it's possible that all you will ever have to carry on your next tour will be a storage medium containing your samples and Programs. Having seen my keyboards disappear into the depths of Stansted Airport to be mishandled by staff who view 'Fragile' stickers as some sort of challenge, I can tell you all about the fear that accompanies the first opening of your cases on foreign soil. A rental company could have provided a Nord Stage locally, but the chances of finding a fully-loaded XP-80 with the correct configuration of SR-JV cards as well as an 18-voice Z1 for hire on an island in the middle of the Mediterranean Sea are as close to zero as makes no difference.

Following the success of the first incarnation of the Nord Stage, the problem facing the big brains at Clavia for the past few years must have been, "How do we keep it new and exciting without fundamentally changing it?". They clearly felt that a cautious approach was warranted because, looking back, its evolution has been glacial. When the original version »

The Rear Panel

The Nord Stage 4 offers four assignable audio outputs, all available via unbalanced quarter-inch sockets. Alongside these, a quarter-inch stereo headphone socket echoes outputs 1+2. A 3.5mm monitor input is also provided, and any signal presented to this is routed to outputs 1+2 and the headphones. Unfortunately, there are no balanced outputs and no audio over USB, the latter of which is starting to look a bit archaic.

MIDI is handled simultaneously by two 5-pin DIN sockets (in and out, but no thru) and USB, the latter of which also provides access for the Sound Manager and Sample Editor applications.

There are six connectors for control pedals. Five of these — Control (a routable expression pedal), Organ Swell (which can also act as a global volume control or emulate the modulation wheel), Sustain (which offers half-pedalling when a suitable pedal is connected), Rotor (which accepts a pedal or a half-moon switch), and a programmable footswitch input — use quarter-inch sockets. The sixth is a DIN input for the Nord TP-2 Triple Pedal, which, in addition to its usual sustain, sostenuto and una corda functions can control things such as layer switching and Program up/down selection.

The Stage 4 88 measures 1282 x 121 x 349mm and weighs in at 19.6kg.

(retrospectively named the 'Classic') appeared in 2005, its piano section was based upon sample memory of less than 256MB, and its polyphony was 40 stereo or 60 mono voices. This was later updated to 500MB in the Nord Stage 2 (reviewed in *SOS* November 2011), to 1GB in the Nord Stage 2 EX (2015), and then to 2GB and 120 voices in the Nord Stage 3 (*SOS* March 2018). Similarly, the original Hammond/Vox/Farfisa organ voicing derived from the Nord Electro was updated to the Nord C2 voicing (minus pipe organs) in the Nord Stage 2, and then to the C2D (including pipe organs) in the Nord Stage 3. Perhaps the largest improvements were to be found in the synthesizer section, which started out as a basic 16-voice polysynth, gained 380MB of sample RAM and two additional voices in the Nord Stage 2, followed by an additional 100MB and another 16 voices with the adoption of the Nord A1-based synth engine in the Nord Stage 3. Given that 18 years have passed, these are not huge leaps, so what can we expect from the Nord Stage 4 — another small increment here and there, or something more substantial?

The Next Stage

I covered many of the Nord Stage's underlying facilities in previous reviews so I'm not going to revisit them here. Instead, I'm going to concentrate on the latest changes to the architecture and the sound engines. So I should start by telling you that, while looking and feeling much the same as before, the Nord Stage 4 is fundamentally different from its predecessors. This is because it has dispensed with the previous two-layer Panel A + Panel B architecture, replacing it with a horizontal architecture that includes two pianos, two organs, three synthesizers, and six complete effects sections, one each for Piano A, Piano B, Synths A, B and C, and Organs A+B together. You can mix the seven instruments freely and, in addition, jump between two Scenes in any given Program. Each Scene has a different combination of instruments switched either on or off so, while switching between the two doesn't change any of the sounds, it allows you to do things such as jump instantly from piano+strings for the verse and chorus to organ+synth for your blistering solo. I would find this hugely useful.

There are numerous other system updates. These start with an increase in the number of Program memories from 400 to 512 and, in addition, the number of Live Program slots has increased from five to eight. If you look at the rear panel you'll also notice that the Nord Stage 4 is able to take advantage of the Nord Triple Pedal 2 as well as the Sustain Pedal 2, both of which add additional functionality beyond the obvious. Another significant improvement at this level is the expansion of the Preset Library functionality, which now allows you to store pianos, organs and synths individually and then drop them into Programs when needed. Finally, the number of fixed split points has been increased from 10 to 11. But don't get too excited by the last of these. Despite the extra option, I have always felt that a handful of manufacturer-defined split points is one of the Nord Stage's most egregious shortcomings, and that the splits for its four zones ought to be user-definable... and I still do.

Turning to the sound engines, the improvements in the piano section comprise a performance compressor to even out your playing (which I'm sure that you don't need but which will no doubt make your FOH engineer very happy) plus three modes of unison that create gentle spatial chorusing but never stray into full-on honky-tonk territory. The major update in the organ section is much more obvious: there are now physical drawbars on all three Nord Stage models (which we'll discuss in a moment). Nonetheless, the LED bar graphs are retained so that you can see the current registration when you recall a preset. There's also the addition of a new B3 Bass Mode with the correct pedal voicing.

Much larger differences have appeared in the synth section because the dual Nord A1-inspired engines in the Nord Stage 3 have been replaced by three layers of a synth that incorporates features culled from both the A1 and the Nord Wave 2. This now offers 1GB of sample memory and the polyphony has been increased to 46 voices but, for me, these are not the most important changes. I'm much more excited by the new mono, legato, high-note and low-note priority modes when using it as a monosynth, plus the programmable arpeggiator that offers monophonic, polyphonic and gated (think 'Baba O'Riley') modes… all of which are on a 'per synth' basis. You might be surprised at the sounds that these arpeggiators allow you to create, with Programs ranging from early Tangerine Dream sequences to basic four-on-the-floor rhythms. When combined with other sections, you'll soon be hearing things that you would never have imagined coming from a Nord Stage. Nevertheless, an Achilles' heel persists in the form of three-stage filter and amplifier contours, which remain stubbornly ADR or ASR depending upon settings. It's such a shame to limit the Nord Stage 4 in this fashion, especially since the Nord Wave 2 has ADSR contour generators. Indeed, I think that it's rather disingenuous of Clavia to claim that the synth in the Nord Stage 4 is (and I quote), the "Nord Wave 2 Synth Engine". I've heard people say that the Swedes can be naughty, but this is not what I thought that they meant.

The last area of sonic improvement lies in the newly multitimbral effects section. This is not just because there are now many more effects units but because the number of available effects has been increased. Many of the new algorithms are extensions of and improvements over previous ones, but I have to draw your attention to the new Pump option in the Mod 1 section, which emulates the nauseating ducking caused by excessive use of dynamics processes. It's a very popular effect, but I hate it with a passion and its very existence (not Clavia's specific implementation of it) makes me angry. But if you have a stronger constitution than I and enjoy doing harm you may be able to tolerate it, and even find yourself nodding along to the rhythm as it pumps and sucks and destroys the continuity of anything vaguely musical. On the bright side, the Reverb section now boasts very useful spring and booth modes plus a cavernous cathedral mode that I like a lot.

Taking To The Stage

The review unit arrived with OS v1.08 installed, so I took a minute (it really is that quick!) to upgrade it to the latest OS v1.12. Having done so, I decided which pianos to delete from the factory set to create the room needed to upload the Bösendorfer Grand Imperial XL and my other favourites from the Nord Piano Library, and then repeated the exercise for the additional Mellotrons, string ensembles and synth samples that I wanted to upload from the Sample Library. This took well over an hour but, once everything was configured to my taste, you could have found me performing all of those unmentionable acts that crumbly old prog-rockers do with nine-foot grand pianos, screaming Hammond organs, Mellotrons and analogue synthesiszers. And, while the playing may have been a bit questionable in places, the sound quality was exemplary.

Of course, you may not want to engage in all of this reprogramming (let alone donning the wig and cape) but that's fine too. I suspect that many owners will be more than happy with the pre-installed sounds and that many Nord Stage 4s will be used very successfully without ever seeing the business end of a USB cable. Indeed, the chances of this may have increased because the additional flexibility has been achieved at the expense of an equivalent increase in complexity. That doesn't bother me — the ability to craft a sound with multiple selections of multiple sound types played through genuinely multitimbral effects sections »

» makes it a far more useful instrument than before — but if you're someone for whom the simplicity of the earlier models was part of their appeal, you'll have to evaluate whether this particular lemon is worth the squeeze. Of course, you could avail yourself of a selection of Nord Pianos, C2Ds and Wave 2s to separate out all of the instruments, and the result of doing so would undoubtedly be simpler to use and look very impressive. But you'd also find yourself significantly poorer and carrying several cases rather than one — not to mention the multiple increase in cables, pedals and DI boxes.

As always, there are three models. The smallest is the Compact, which sports a 73-note semi-weighted waterfall keybed. If your primary need is for organs and synth sounds, this is probably the most appropriate for your needs. Next comes the 73, which offers a 73-note hammer-action keybed. I fear that this model is a mistake; emasculating the previous HP76 by three keys will be acceptable for some uses, but it's an unnecessary restriction when playing acoustic pianos and also reduces your ability to create useful splits. The third is the 88, which, for me, is the only choice if you're expecting to play lots of pianos on the Nord Stage 4. It's the right width, and its keybed appears to have been upgraded to the triple-strike action found on the Nord Pianos. Perhaps the feel isn't quite as nice as the top-of-the-range Kawai keybed installed in the Nord Grand, but I found it to be very pleasant and playable. And, despite all of the changes, the sizes and weights of each of the models are much the same as their earlier equivalents, which makes the whole range very manageable without them ever feeling lightweight or flimsy.

All The World's (Using) A Stage?

Looking back, the longevity of the Nord Stage series has been remarkable. What makes it all the more so is that Clavia have refused to let older models die. Rather than tell existing owners to dump their Classics, 2s and 3s onto the secondhand market and hand over large wodges of cash for the latest whizz-bang, sell your granny, you-can't-survive-without-it Stage 4, the company have continued to support and upgrade the earlier versions. Indeed, the Classic was updated as recently as 2018 and — within the limits of what's possible on its older hardware — is now compatible with the newer sound libraries, Sound Manager and USB drivers, as well as hosting improved effects. Similarly, the latest updates for the Stage 2 and 2 EX appeared in 2019, and for the Stage 3 in 2021. I think that existing owners owe Clavia a huge 'thank you' for this; developing free updates is far from cost-free for the manufacturer — it takes significant time, facilities and effort. So if the initial outlay for a Nord Stage is higher than you might expect (or want), ask yourself how much you value a manufacturer's commitment to keeping your pride and joy current for 15 years after it's been superseded. Having said all of that, the Nord Stage 4 is a huge step up from previous incarnations, and I suspect that there will be many owners who will seriously consider upgrading to it. I wouldn't blame them; the Nord Stage 4 is so far ahead of its predecessors that I doubt I could ever go back.

The thing that will cause them — and potential new owners — to pause for thought is the cost. At almost $5700 (or, to put it another way, nearly twice the price of a K2700 or a nearly-new Kronos 88) the Nord Stage 4 88 is almost as far from cheap as a current digital keyboard can be. What's more, were I ever able to make one the primary instrument in my rig, there would undoubtedly be occasions when I would find myself bemoaning its deficiencies in comparison with those powerful workstations. But I would never find myself cursing its playability or the sounds and inspiration that I get from it. It's a beautiful instrument to own, to play and to hear, and I have no doubt that it will be every bit as popular as its predecessors.

$ Nord Stage 4 Compact 73 $4899, HA73 $5399, 88 $5699.
W www.nordkeyboards.com

The Sample Editor

The Nord Sample Editor 4 was updated in February 2023 to coincide with the announcement of the Nord Stage 4. It's compatible with Windows 7 onward when used with the Nord USB Driver v3.00 or later, and with revisions of macOS from v10.10 onward.

It's a very slick piece of software. Just record audio within the Editor itself or drag and drop audio into its main window, edit its attributes — start, stop, loop, pitch, level, key mapping and so on — and then, when happy, click on 'Save and transfer to Nord'. If you want to map a single sample across the whole keyboard it takes just seconds; if you want to set up a complex multisample it obviously takes a bit longer. Once transferred, the sample is immediately available on the keyboard and you're free to treat it as you would any other underlying waveform, applying filtering, contouring, effects and everything else on offer.

The sample in the image here is part of an extended sound that I created some years ago using the vocal mangling capabilities of my V-Synth XT, and it sounds like nothing that you would ever expect to obtain from a Nord of any flavour. But it took me less than a minute to trim, loop and transfer it, whereupon the Nord Stage 4 suddenly sounded like a V-Synth. Don't underestimate the power of the Sample Editor — it greatly extends the palette and flexibility of the big red beastie.

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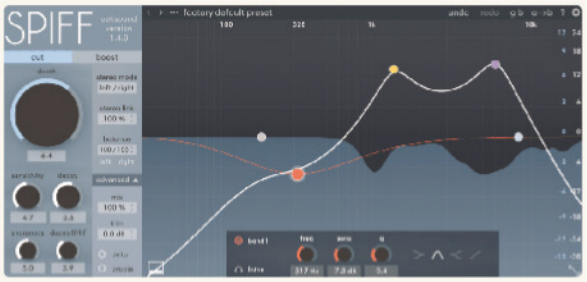

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Boss GM-800
Guitar Synth

It's nearly half a century since the first guitar synth became available commercially — and, boy, have they come on a long way since then!

PAUL WHITE

Boss's GM-800 guitar synth may look a lot like their SY-300 — both models have roughly the same form factor, colour and general control layout — but they're very different beasts. Based around the latest generation of pitch-tracking technology, the new GM-800 follows the paradigm of the latter-year GR series, specifically in that it uses a divided guitar pickup to send the individual string signals into a digital signal processor that extracts the pitch and velocity information from the notes before using it to trigger the kinds of sample-based sounds you might expect to find in a keyboard synth. The GM-800 can also output MIDI to control an external MIDI synthesizer or sound module, and accepts MIDI in, not only for deep parameter control but also to allow it to double as a MIDI sound module itself.

Boss GM-800
$749

PROS
- Fast and accurate pitch tracking.
- ZEN-Core compatibility.
- Integrates well with computer DAW software.
- Sensibly priced.

CONS
- Older GK pickup systems require a converter box.
- Sound transfer from Zenology a bit clunky (but may yet improve).

SUMMARY
The GM-800 combines enhanced pitch tracking with access to a wide range of sounds at an affordable price.

Overview

Before going further it should be noted that the connection system between the pickup system and the GM-800 no longer uses a 13-pin DIN connector. Instead, it uses a serial communication system that works with a standard TRS jack cable. The new compatible pickup is the GK-5, but for those with existing GK pickups or guitars with compatible pickups built in, an adaptor called the GKC-AD converts from the older system to the new one. This takes the form of a utilitarian metal box with a 13-pin GK input, a serial GK output and a separate jack output that carries the standard guitar sound. Status LEDS show power and connection; power comes from an included inline adaptor. Until a connection is established between the GM-800 and the GKC-AD, a green LED flashes but remains solidly lit once a connection is made.

Setup, accessed via the Menu and Page buttons, includes fine-tuning your GK pickup options. First you select which type of pickup you have — this may be a third-party instrument, including those with piezo divided pickups — then you need to adjust the sensitivity for each string so that they all produce a similar output for a given picking force. There's a long list of pickups from which to choose, including third-party systems by Graphtech, Fishman and more.

It's very important to get the pickup setup right, as this will affect how well the instrument responds to your playing style. Set the sensitivity too high and not only will you lose out on some dynamics when picking hard, but there's also an increased risk of triggering unwanted sounds from fingering noise. On the other hand, if you set it too low, you might not be able to reach the maximum level even when picking hard. To help with setup, a set of bargraph meters show you how much output you get from a string as you pick it. The general aim is to ensure that your normal hard picking gets the meter close to maximum without actually reaching it. There are also setup options for selecting guitar or bass, the latter requiring a GK-5B pickup or an earlier GK bass pickup plus a GKC-AD adaptor. As some users may have more than one GK guitar, its is also possible to create and save several different pickup settings.

The output can be mono or stereo and there are internal settings to voice the sounds for either a regular guitar amp or a full-range speaker system (such as a keyboard amp or PA feed), which is of course preferable. The left output also doubles as a headphone output and in addition to the 5-pin MIDI in and out ports there are two TRS Exp jacks for connecting switches or pedals. The unprocessed guitar sound has a separate output on an unbalanced jack.

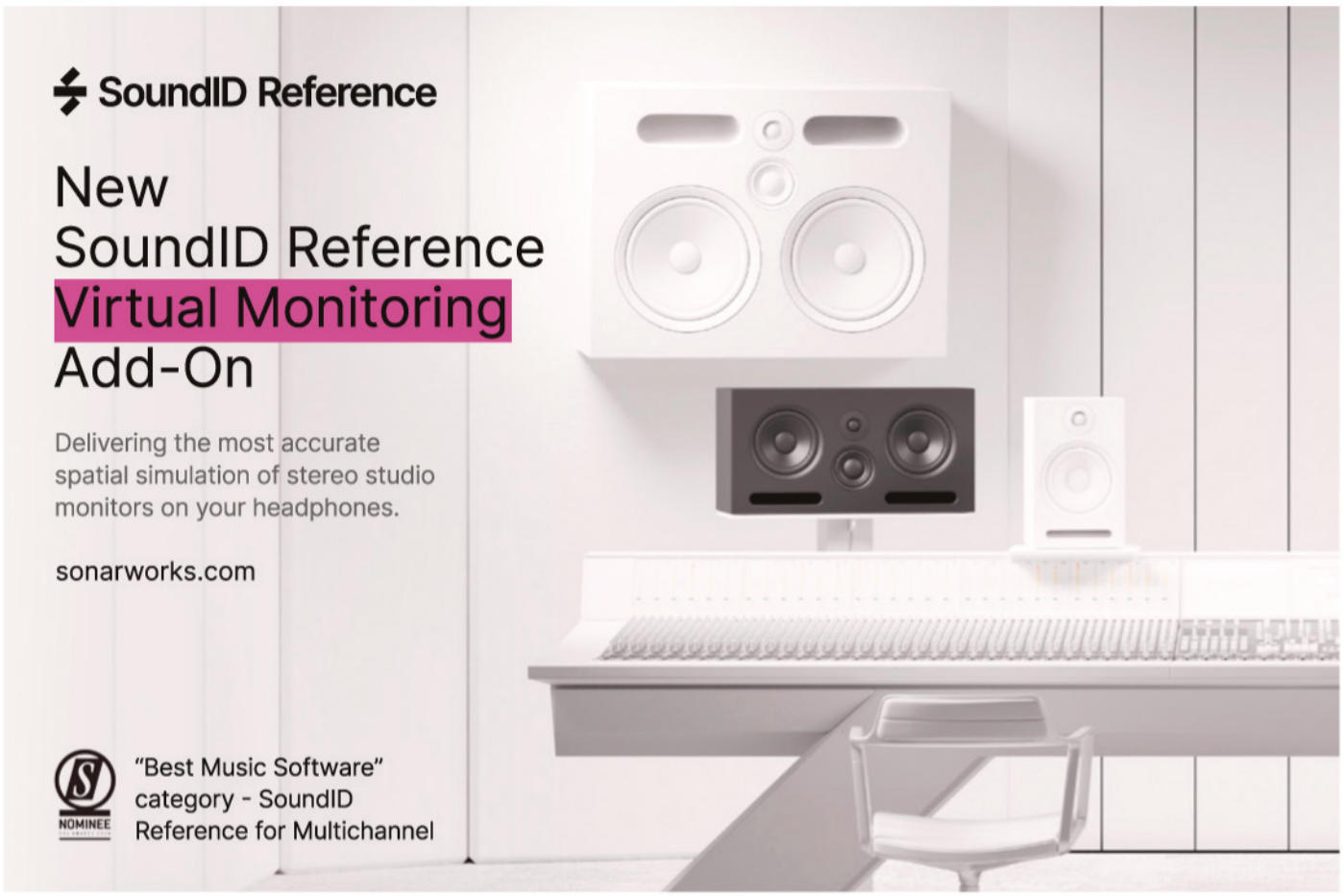
The Shape Of Progress

Although physically small, the GM-800 has many advances over its predecessors, the most notable being its fast and accurate pitch tracking. It has a USB port for connection to a computer and while it wasn't available for my review tests, I'm told that an editor will be available 'imminently'. The GM-800 can also record directly into a DAW via USB and includes the option to record the output from the divided pickup onto your DAW audio tracks, in such a way that the performance can later be replayed to trigger the sound generator of the GM-800. This opens the door to experimenting with different sounds after making your recording. A second USB port accepts a memory stick, used for firmware updates, additional patch storage and patch transfer.

Importantly, the GM-800 is a ZEN-Core compatible device. The Zenology plug-in and additional sound packs can be downloaded from the Roland Cloud using the USB connection. Essentially, you install Zenology as plug-in instrument (this has a load of core sounds before you even buy any expansion packs) and then, from the plug-in, you can export your chosen sounds to the desktop and transfer them via a memory stick to the GM-800 hardware. The process is a little convoluted, but you can transfer a large number of sounds in one go, so it is well worth doing. A GM-800 driver must be installed on the host Mac or Windows computer, and for Mac users supported operating systems are from macOS 11 onwards.

Operation

Dominated by a large display that shows the patch number and name when in default Play mode, the control panel is designed to be non-intimidating. There are four different views for Play mode accessed using the »



The hardware control interface is pretty intuitive, with the four encoders below the main screen controlling what's displayed above.

Page buttons so you can also opt to view the four Tone names or the functions assigned to the footswitches for the current patch (or 'Scene', as Boss refer to it). Other than the Output Level control, all the knobs are rotary encoders, the four below the display accessing what's shown above. In Play mode these can be used to change the balance of the four Tones making up the sound. Sounds can be built up using up to four layers of sample-based sounds (Tones) plus there's a further layer of rhythmic elements such as drum kits. There's also an onboard selection of effects such as delay, reverb, modulation and so on, some being editable in surprising depth.

The Select encoder at the top right moves through the patches when in Play mode but has other select functions in setup mode or when editing sounds. Below are buttons for Exit, Write and Menu plus left and right Page buttons. Menu is where you find the setup details and access to editing. Illuminated buttons at the left of the display are available for the four Parts, with a further button turning the Rhythmic component on or off. When in Parts mode, selected by pressing one of them, the four buttons can be used directly to switch parts on or off and bring up a screen display allowing Tones to be selected and modified as well as accessing MFX (effects), String and Others settings such as Mono/Poly, Legato, Chromatic and Hold functions.

Scenes store the Tone layers, their settings, any effects used, control configurations and also other details, such as whether individual strings have been muted or not. The opportunities for sound editing are fairly basic so as to keep everything guitarist-friendly, so there's access to the essentials such as envelope shaping, vibrato, legato mode, portamento, filtering and effects, but nothing scarily deep. You can edit some of the effects in great depth if you are that way inclined but in most cases just tweaking the main parameters is all that's needed.

A pair of Up and Down footswitches can be used to navigate the Scenes while the Control One and Two footswitches, which can operate either as latching or momentary, handle Scene-specific functions, such as switching some of the Tones on or off or instigating a control change. Each footswitch has an associated rectangular LED.

Playability & Sounds

As shipped, there are 100 sounds ready to go and, as you might expect from a factory sound selection, there's a bit

There's extensive rear-panel connectivity, including both ins and outs for the pickup and MIDI, and USB ports for connection to a computer and to host solid-state memory devices such as USB pen drives.

Studio Essentials.

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SOUND BETTER
BETTER SOUND

The GM-800 launch was accompanied by that of a new GK pickup system that uses a serial connection over a standard TRS cable. The pickup is available separately, and you can also buy adaptors for older pickups.

of everything here, from orchestral and acoustic instruments to organs, pianos, EDM and bass sounds. There are also numerous drum/percussion rhythms — not every style of music will need them, but if you want to be a one-person EDM band... The quality of sounds is excellent but inevitably only a small number of the factory Scenes are likely to fit in with your own musical leanings. Fortunately, then, it's easy to create new sounds simply by switching out the existing Tones (samples) for something more suitable. This is easy from the front panel, but the free macOS/Windows Boss Tone Studio editor software makes it easier still, as well as making the deeper editing controls more accessible. There's a wide range of Tones, arranged by instrument

By the time you read this, a Mac/Windows control app should be available — this screenshot is from a beta version shared with the author.

type and with a range of variations in each category. If you subscribe to the Roland Cloud or have purchased Zenology separately as a plug-in after registering for a free account, you'll also have a wealth of sounds available to populate your GM-800.

Splits can be arranged either by string (eg. bass sounds on the lower two strings and something else on the remaining four) or as you move up a string set by fret position. You can also arrange cunning layered sounds by, say, putting a bass sound on the bottom two strings, a pad sound on all the strings and adding a mono lead synth set so that one of the control switches brings in a little vibrato just to that voice. This really does create the illusion of three synths playing at the same time!

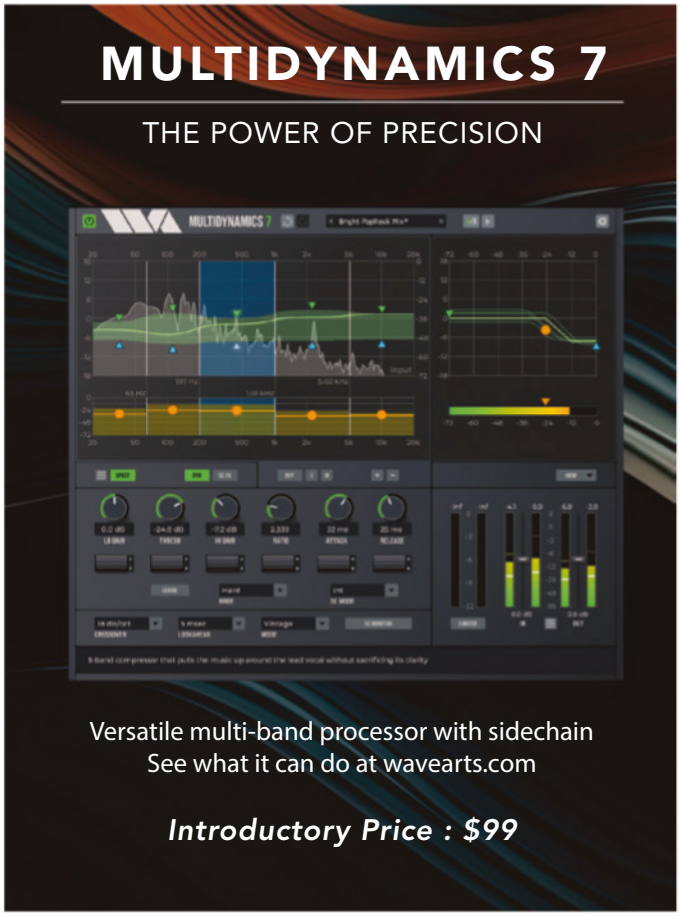
When it comes to playability, an inherent limitation with guitar synths is that, unlike keyboard synths that will sustain a note for as long as you hold a key, guitar string vibrations eventually die away to the point that no useful pitch data can be extracted. Yet, the GM-800 can track the string pitch for a surprisingly long time and you can further extend a note by setting a longer release time so that the note continues to sound and to fade away after the guitar note has been stopped. On earlier guitar synths, this could result in erratic pitch droops, especially in pitch-bend-follow rather than chromatic mode, but here things seem much more solid. There are also Hold pedal functions that can be brought into play so there are plenty of options for keeping the sound going.

The new tracking algorithm definitely seems better at discriminating between a deliberate pick attack and random finger noise, which means there are fewer unwelcome 'squawks'! If you accidentally strike a pinched harmonic, you may be greeted by a brief chirp or pitch swoop, but on the whole the playing feel is very natural with pitch bends and vibrato (not in chromatic mode of course) feeling as natural as when bending conventional guitar sounds. Instruments such as pianos are best set up using chromatic mode, of course, since pitch bending is not a normal part of their sound, and the same is true of most synth pad sounds — chromatic mode will tighten up any minor tuning discrepancies, and you can set up the control switches to bring in vibrato or pitch glides as necessary. As with any guitar synth, slow-attack sounds are the most forgiving of messy playing, while piano sounds and the like require a little more care if unintentional 'jazz notes' are to be avoided.

Summary

Given its realistic price and wealth of onboard sounds, the GM-800 makes a very attractive proposition for anyone wanting to add a guitar synth to their armoury. It boasts some fairly significant technical advances and definitely feels more natural to play than previous systems. Unless you have an existing GK pickup (in which case you'll need to buy a GKC-AD adaptor), you'll need to buy a GK-5 pickup separately, which adds to the cost, but you still get a very powerful system for significantly less than I paid for my GR-500 back in 1977!

$ GM-800 $749. GK-5 pickup $249.99. GK-5B pickup $299.99.
W www.boss.info



IK Multimedia UNO Synth Pro X

Synthesizer

IK Multimedia's new UNO Synth Pro X is a statement of intent.

WILLIAM STOKES

The last iterations of IK Multimedia's UNO Synth series saw the Italian company's paraphonic UNO Synth Pro and UNO Synth Pro Desktop commit to the trajectory set in motion by the first UNO Synth, throwing down the gauntlet as serious contenders in the analogue synth game. In my review of those instruments back in late 2021, I noted that they were surprisingly powerful, that their onboard effects were surprisingly good, and maybe the takeaway from this is that we just shouldn't be surprised anymore. In any case, I was therefore not surprised, upon unboxing the UNO Synth Pro X, to see that IK have demonstratively sought to up their synth game and extend the appeal of the series 'upwards'; in other words, this time around it feels like the company have, you might say, more 'serious' synthesists in their sights. Make no mistake: these are quality electronic instruments by any standard, even if their names do sound like a jumble of all the marketable terms IK could think of. Including the word 'synth'.

I say 'more serious' carefully, because that doesn't necessarily mean 'better' — Teenage Engineering for one blew that notion out of the water long ago with the gloriously dumb yet generation-definingly good OP-1. It's by no means esoteric,

IK Multimedia UNO Synth Pro X
$499

PROS

- Hugely improved panel layout.
- Excellent range and depth of modulation.
- Intuitive and flexible sequencer.
- Quality effects and drive.
- Oodles of preset storage with independent sound and sequence recall.

CONS

- 'Bassline' mode feels a touch superfluous.

SUMMARY

IK Multimedia have shown just how serious they are about their synth game with the greatly improved UNO Synth Pro X: a characterful, classy and highly functional instrument that will prove as valuable in the studio as it is on stage.

but nonetheless a sense of accessibility doesn't seem to be at the top of the UNO Synth Pro X's agenda, at least in the way it was with the UNO Synth Pro. For one thing, the Synth Pro came in two iterations: a Desktop version (with a form factor exactly the same size as the Pro X), and a keyboarded version that I should point out boasted a very nice-feeling Fatar keybed. With the Pro X, IK Multimedia have forgone the keyboard and committed to one desktop version, presumably on the assumption that anyone wanting that dimension of playability will have their own MIDI keyboard. No hand-holding on that front, then.

Far from its predecessor's panel's minimal interface, ostensibly consisting of four assignable knobs and grid of parameters, the Pro X's matte-finish panel is replete with controls. It's bold and upfront with its functions, many of which are carried over from before, though there are a raft of additions and improvements to speak of as well. It's not far off a knob-per-function layout, and those used to conventional subtractive synth architecture will find themselves on ostensibly familiar ground.

Logic Pro X users may raise eyebrows at the name — as we've come to expect from IK Multimedia the Pro X is set to pair famously with your computer, with USB-C connectivity and a brand new deep-diving digital editor that can also assume software plug-in form (side note: you can't transfer presets from the Synth Pro to the Pro X). But as far as I can tell there's no particular persuasion toward Apple's DAW. In fact, in one interview IK product manager Enrico Dell'Aversana lightheartedly referred to the 'X' as standing for 'experimentation', and in fairness it's easy to see why. The Pro X's interface invites sonic exploration in a manner we've not seen from the Synth Pro series up to now, and it's streamlined many features to make the whole process that bit more fluid and enjoyable as well. It's also worth noting that the Synth Pro X's price is the same as that of the keyboarded Synth Pro, and significantly higher than the Synth Pro Desktop. The message is clear: IK Multimedia mean business.

Transport Is Arranged

Users of the Synth Pro Desktop will notice that the Pro X has changed things around a little — or a lot — when it comes to panel priority. Where the Synth Pro had a two-and-a-half octave capacitance-sensing keyboard taking up at least a third of the overall panel space, with a miniature sequencer squeezed above it, the Pro X priorities its sequencer with generously-sized buttons and pops a miniature single-octave keyboard just above. This is a good move — not least because I don't think anyone will miss that keyboard (I'm yet to see a single person ever actually play a capacitive keyboard musically), but also because it emphasises the value of the Pro X's sequencer as central to its character and playability. In many ways it serves to render the Pro X more a *bona fide* sequencing instrument in the tradition of Elektron or Roland (more on that anon) than a keyboard-controlled synth that happens to have a sequencer included. The little keyboard is still useful for programming sequences and auditioning sounds, of course, but I'm of the view that on an instrument this size you really don't need any more than that. I also feel the Pro X's look and layout brings it into a more Korg-ish territory, perhaps because of its faint aesthetic shades of the Electribe or Drumlogue.

The UNO Synth Pro's rear panel hosts an input for an external power supply, a USB-C port, 3.5mm sockets for CV/gate, audio input and headphone out, a pair of quarter-inch audio outputs and full-size MIDI I/O ports.

» Sequences of up to 64 steps can be punched in step by step, recorded in real time with the keyboard (the Synth Pro's recording metronome is carried over to the Pro X) or even imported from the 10-mode arpeggiator. Usefully, when loading up a preset the Pro X offers the choice of loading the sound, the sequence, or both, so you don't need to recreate any sequences to play them with different voices. There's more here than I have space to cover, but it's a big tick in the 'sequencer' box.

Engine Room

The left half of the panel is taken up by the Synth Pro X's three oscillators and dual filters, building on the design of the Synth Pro (which itself greatly expanded on the original UNO Synth's dual oscillator, single filter design). It can operate in paraphonic, legato or mono modes, selectable via a Voice button in the top left corner of the panel. This is as good a time as any to say I'm a big fan of paraphony, particularly when it comes to sequencing: it maintains the limitation of a monosynth and its emphasis on sound sculpting, but still allows the odd unexpected chord to appear in a sequence, or the occasional overlapping of notes in a melody line.

Singular knobs for wave morphing (from triangle through saw to square and, finally, pulse), tuning and level are switchable between oscillators 1, 2 and 3 with a button that can also be held down to edit all three together. As on the Synth Pro, to my ear the oscillators don't have buckets of character, but they sound very nice and very clean. The two filters, meanwhile, each get their own panel controls for cutoff, resonance and envelope amount. Filter 1 is OTA-based, while filter 2 is SSI chip-based. A column of buttons allow for a range of various modes: you can link the two to be controlled by filter 1's controls, switch filter 1 between low- and high-pass modes, switch filter 2 between 2- and 4-pole modes, and put the two in either parallel or series in the signal chain. Right out of the gate there's a real emphasis on the filters as the Synth Pro X's primary sound sculpting tool; much more than on the Synth Pro. It's a bigger and more detailed section than that of the oscillators, even if the oscillators do also offer FM, sync, noise and ring modulation.

Ring modulation? Dual filtering? Of course the Korg MS-20 springs to mind. No complaints here, of course; the MS-20's pair of filters is one of the reasons it has a place in the hearts of synthesists the world over, and I was happy to see its architecture nodded to here, since it's surprisingly rarely seen in other instruments. The Synth Pro X's filters don't have the same analogue snarl as the MS-20's in and of themselves, but as a pair there's something rather brilliant about them — much more than the sum of their parts. It's also worth pointing out that the Synth Pro also had dual filters, but it goes to show how much interface design can influence a player's approach: on the Synth Pro it felt more like a handy bonus feature. This time, with the two ready and waiting on the panel for hands-on tweaking, it feels positively crucial to the workflow. I will also say that they magnificently work to channel the filter of the now-legendary TB-303, and were more than capable of imparting the kind of tasty acid house throatiness I was hoping for from them.

This is not lost on IK Multimedia, who cite the 303 as the inspiration behind one of the Synth Pro X's more intriguing features: Bassline mode. This mode makes a few nudges across the board to alter the Pro X's character in the name of fluid low-end sculpting, for instance reducing the upper frequency limit from 20kHz to 5kHz and limiting the envelope to just the attack and decay stages. It also enables an Accent feature on the envelope for emphasising certain steps in a sequence. While this mode certainly does a brilliant job of channelling oodles of watery, distorted 303 energy, one frustration came with the fact that, at least on the loan unit in front of me, Bassline mode doesn't seem to be a simple on/off affair: hit the button to enter the mode; hit it again and you'll exit the mode but with a smattering of parameters changed, with no immediately clear way of going back to where you were. Perhaps this will be addressed in [*cough*] a future firmware update? Part of me feels that this kind of bass-building is already inherent to the Synth Pro X's character; surely it could have been worked into the overall workflow (via a button to remove the latter two stages of the envelope, for example) instead of being spoon-fed via its very own global mode. It's a fun feature, though, to be sure, and at the end of the day there's no mandate to use it.

In The Deep End

The top right of the UNO Synth Pro X is home to a small OLED screen and grid of master control buttons for preset storage, tempo adjustment, arpeggiator toggling and more. Happily, with the panel of the Pro X providing so many controls at my fingertips I found myself deferring to the screen and its encoder surprisingly little — which isn't something I could say of the Synth Pro. In fact, Enrico Dell'Aversana has previously chalked up the Synth Pro as a 'preset machine' compared to the Pro X. Don't look at me — his words, not mine — but I will say there's far less dreaded menu diving to speak of this time around. What the screen does do very nicely is display values, allow for global settings, enable preset naming and recall and the like. At the end of the day, I'd say IK Multimedia have got the screen-to-panel balance about right.

To the left of this section is a single envelope generator, switchable via a button between amplitude and filter control. Said button also allows for the editing of a handy third envelope, which can be assigned to go almost anywhere. Another very nice touch, again rarely seen on an instrument like this, is the option to have the envelope's attack and decay stages loop like an LFO. Beneath this is the LFO proper, with a single set of controls for both LFO 1 and 2; once again switched between via a button. Frequencies can be freeform or sync'ed to a clock division at the push of a button, which is another nice, sequencer-friendly addition, and there's also the option for the LFO to fade in with

a truly languid maximum fade time of 10 seconds.

When it comes to modulation at large, I was extremely impressed by the Synth Pro's capacity for incredibly deep and flexible routing. Three unassuming Source, Destination and Amount buttons in a section labelled Matrix on the far left of the panel allow for slick, upfront modulation programming. Push the Source button, turn a knob to assign; push Destination and turn another, then adjust the amount using the Pro X's data encoder. It's quick and phenomenally creative, not to mention open-ended. Parameter changes can also be recorded in real time across a sequence, and there's even step-by-step parameter 'locking' for truly articulate, synth-wide sequencing with automated parameters and effects.

Effective Communication

As with the Synth Pro, the effects on the Pro X do a good job of punching a fair way above their weight. The three effect categories of modulation, delay and reverb are squeezed into the same four-knob section beneath the LFO controls, with — you guessed it — a button to toggle between them, while the post-filter, twin-diode drive circuit is positioned on the other side of the panel beside the main volume knob. This I'm an advocate for: one-knob analogue drive is never unwelcome, and for me it almost always does something to reiterate the joy of using hardware and bring out some real character. While I would have loved the option to place the drive pre- or post-filter — which doesn't feel like a stretch given the amount of flexibility on offer here — the Pro X's drive nonetheless sounded great and gritty to my ear, adding genuine value to the overall result. Further, placing it alongside the volume knob makes a great pairing for broad-strokes gain and tone control; between the two of them there's a lot of scope.

Meanwhile the other three effects, as mentioned, are grouped. Their shared controls aren't the most intuitive, with labels for the reverb effect (Time, Size, Filter and Amount) given priority on the panel over those of the other effects for some reason. In fact, I would probably have preferred the knobs to simply be labelled 'param 1', 'param 2', etc, so I could know from the outset simply to reach for the screen and data encoder for all my info, effect-wise. As it is there is a lot to choose from with the Synth Pro X's effects, but some scanning of the manual is almost certainly needed to figure out how to access it all. Shift-Amount, it turns out, is used to choose from a host of sub-categories per effect. Reverb offers a choice of hall, plate or shimmer. Delay offers mono, stereo, doubler, ping-pong or LCR (a mode that cycles the delay signal from the right channel through the centre and into the left channel). Modulation effects have been given a reboot: there's chorus as before, but the Pro X's eschews the phaser and flanger offered by its predecessor and instead opts for a uni-vibe effect. Given the option, I would probably have preferred the previous effect selection, but that's a matter of preference. There isn't much to complain about here; the OLED screen means things don't take too long to adjust and at the end of the day the effects sound very good, particularly when one starts to bring modulation into the picture.

More Than The Sum Of Its Parts

On that note of modulation, all things considered it's the sheer depth of features like this that really constitute the heart of the Synth Pro X. It certainly sounds good in a static mode of operation, but when those two filters start to move and grind against one another, when waves morph to and fro, when effects swell and die down across a sequence, it all amounts to something rather special. What gives the UNO Synth Pro X its edge isn't so much its raw sonic power — as I've said, its oscillators didn't stagger me but they certainly get the job done, and the same goes for its filters. Its functionality, however, is excellent. Suffice to say: three oscillators, dual multimode filtering and such a deep modulation matrix on a synth this size — and at this price — is impressive, and to further furnish it with some quality effects and analogue drive is laudable. I mentioned that the Pro X feels aimed at more serious synthesists than the Synth Pro, but at the same time I don't believe IK are trying to placate purists here. The UNO Synth Pro X is a powerful and multifaceted instrument that will reward hours of experimentation, yet it still manages to stay fun and light on its feet in all the ways that matter. Explorative enough for the studio and upfront enough for the stage, it's very much a success for IK Multimedia.

$ $499.99
W www.ikmultimedia.com

Going Soft

The UNO Synth Pro X welcomes a brand new software editor, building on the one that came with the Synth Pro. While it's not possible to transfer presets from the Synth Pro to the Pro X, it has become a lot easier to edit and store sounds thanks to an updated interface that both mirrors and expands the hardware unit's panel. The workflow is therefore nicely transferable between the two, while oscillators, envelopes and more are rendered in a knob-per-function layout, with automated parameters moving in real time. Its modulation matrix and effects page also break out a host of the Pro X's functions and present them on a platter, while the librarian page allows for easy storage and management of presets. The Pro X can be bus-powered, so I can easily imagine it becoming best friends with your laptop; entailing all the flexibility and portability of working in software while sculpting sounds in the analogue domain.

Soundghost Scatter

Granular Delay Plug-in

Scatter is a very simple-to-use granular delay that generates up to 64 grains with randomised positions, with stereo placement based around a fixed one-second recording buffer. At lower grain values, typically below 10, you get some nice 'tinkly' effects, especially if using the pitch-shift function, whereas at higher grain values, the sound takes on a more dense, shimmery character. The pitch-shifting function adds a lot of interest to the sound, especially if pitch randomisation is used. Pitch changes can be locked to chromatic, octaves, fifths or major/minor scale interval modes with a maximum pitch range of two octaves up or down. Scatter is available in AU (for macOS 10.12 and up) and VST3 (for macOS 10.12 and up and Windows 8 and newer) formats.

An onboard reverb adds density while a Freeze 'snowflake' button allows the current buffer contents to be replayed indefinitely until released, but you're still able to experiment with the controls to manipulate the frozen sound. Grains may be output in a forwards or backwards direction, as set by an arrow to the right of the grain display. The grains slider, just below the grains display, sets the number of grains from zero to 64 in one-grain steps. As the plug-in runs, the dots in the grain display reflect the number of grains selected and wink in and out to show grain activity. The brightness of the dots reflects the amplitude of the grains. Each of the four rows of 16 dots has a different colour.

The main controls are set out as knobs. Size sets the length of the grains, from 30ms to one second, while Feedback adjusts the amount of granular signal fed back into the recording buffer, generating longer decay times. Space brings in a big-sounding reverb while Shape provides choices of grain shape (rectangle, exponential decay, fast decay or smoothed triangle). Mix sets the wet/dry balance while Pitch determines the pitch of the grains according to the intervals of the currently selected transpose mode, selected via a drop-down menu.

The outer ring of the Pitch control sets the degree and direction of randomisation within the selected pitch mode, where the range of pitch movement is shown by blue shading within the ring. The options are up, down or both, set by clicking on the two arrowheads to the right of the Pitch knob. Using very low grain density with a clean (ideally root) note stored in Freeze mode, the major and minor pitch modes can be used to conjure up some very useful random melodies.

While not as flexible as some of the more costly 'big gun' granular delays, Scatter is much simpler to use and can add a lovely ambience to an otherwise bland sound, simply by mixing in around 10 percent of the wet signal. The effects range from a sprinkling of characterful tinkles and rumbles to a dense cloud of crystalline complexity and, unless you are very heavy handed, most of what Scatter serves up is musically useful. I tried it on pad sounds, eBow guitar and simple melody lines, and in all cases it added something very appealing. Its ability to create something new from frozen sounds is also worthy of exploration.

Paul White

$ $49 (about $60). Reduced to £39 ($48) when going to press.
W https://soundghost.net

Hughes & Kettner StompMan

Guitar Amplifier Pedal

The StompMan may look like 'just another drive pedal' but actually it's a fully-fledged 50W amp head that can fit on a pedalboard. (The 50W rating applies where a 4Ω speaker is being used; 8Ω and 16Ω speakers would limit the power to 25 or 12.5 Watts, respectively). It can also be used for re-amping guitar parts that have already been recorded or as a power stage for an existing preamp. The 'amp in a pedal' idea isn't new, but Hughes & Kettner have earned an enviable reputation for top-tier guitar tone over the years so I was keen to see where they'd taken it.

The StompMan's analogue preamp circuitry feeds a Class-D output stage, with power coming from an external 24V, 2.5A switch-mode, laptop-style power brick. When powered up an orange glow is visible through the slots between the two footswitches. Based on Hughes & Kettner's Spirit Tone Generator technology, the StompMan is designed to work well with other pedals and includes a serial effects loop on send and return jacks so that effects can also be placed between the preamp and power amp. The send jack also doubles as a line output if needed, and the signal delivered here includes the saturation element of the power stage. There's no built-in speaker emulation, but feed a separate speaker emulator device from this line out and you have an effective DI recording source or PA feed.

In addition to a footswitch for bypassing the effects loop (or the preamp section, if that mode is selected using a rear-panel recessed switch), there's a Boost footswitch that increases the gain by up to 6dB, as set using the Solo volume knob on the rear. In preamp bypass mode, a trim pot on the bottom of the case allows the gain structure to be adjusted to match the source. Another recessed switch activates auto energy saving mode so that the amplifier puts itself to sleep when unused for more than 90 minutes. It powers back up when a signal is detected.

The top-panel controls have been distilled into Tone, Resonance, Presence and Sagging, in addition to the expected Gain and Master volume controls. Tone isn't a simple filter but rather moves from

a scooped US amp feel at one extreme to a more UK amp character with mid emphasis at the other, offering a neutral sound in the centre. So if you need a softer tone, then backing off the guitar's tone control achieves that. Sagging emulates the way a valve amp's power rails drop when the amp is being pushed hard, producing a type of compression and adding harmonics, but here it can be dialled in at any volume. The Gain and Sagging controls work together in shaping overdrive tones, while Resonance adds weight to the bass end by emulating the low damping factor of a typical guitar amp and speaker setup. In practice this endows the low end with a solid sense of authority but without making it sound boomy or woolly — very much like a valve amp, in fact. When bypassing the preamp to use the StompMan's power amp with a third-party preamp, the Sagging control remains active for adjusting the power amp character.

There's enough gain range to serve up both clean and moderately overdriven tones, though using an external drive pedal or separate preamp pedal effectively turns the single-channel StompMan into a two-channel amp and allows those players who need more extreme distortion to achieve it using an appropriate pedal.

The StompMan's on-paper promise was all borne out in practice. Sonically, I was immediately impressed by the StompMan, especially when dialling in those elusive on-the-edge blues tones, which it handles exceptionally well with the right amount of touch responsiveness. At near maximum Gain and three quarters Sagging, my Strat (neck pickup) got very close to the jangly 'Hey Joe' Jimi Hendrix sound, with the character of the guitar remaining intact. Backing off the Gain also delivered a solid clean sound with loads of headroom. If you need more overdrive, then hooking up a pedal works just fine.

I did find that, in the last quarter of its travel, the Sagging control caused some noise as the pot was turned, and the background noise of the amp also increases at maximum Sagging settings. However, the control does produce a realistic sense of amp compression and the aforementioned background noise is no worse than you'd hear from a conventional valve amp, and in most cases I got good-sounding results at lower Sagging settings (so noise wasn't an issue).

The StompMan's ability to work well with pedals and to provide a big sound at low volumes makes it a very practical studio amp, as long as you have a decent speaker cab. My tests were done with a small 1x12 cab but this amp can drive anything within its impedance range and, when cranked, it is extremely loud. Having an effects loop is a big bonus, not least because reverb and delay often sound best coming after the preamp distortion, and for re-amping there's the choice of going through the whole signal chain or just the power amp stage. And as the power brick has a standard IEC mains »

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» socket at one end, it can be left attached to your pedalboard.

In summary, the StompMan makes it easy to achieve very high-quality guitar sounds with a minimum of effort. It also achieves a satisfying playing feel, which is not always true of other solid-state designs or modelling solutions. Its feature set makes is equally useful live or in the studio and it is attractively priced. I'm starting to get that levitating credit card feeling! *Paul White*

$ $239.
W www.yorkville.com
W http://hughes-and-kettner.com

Strymon Ultraviolet

Uni-Vibe Emulation Pedal

The Ultraviolet is the third pedal to join the ranks of Strymon's new compact line of ARM-powered processors, all of which include Class-A JFET preamps and which are equally happy accepting guitar or keyboard signals. TRS jacks are used for the audio connections, allowing for mono or stereo operation, and the pedal can be set to true or buffered bypass.

The original Shin-Ei Uni-Vibe used an LFO to modulate a light source, around which were located four photocells: its operation was somewhat like that of a phaser, but differences between the four stages gave the Uni-Vibe a unique pulsing sound. Artists including Dave Gilmour, Jimi Hendrix and Robin Trower made great use of the original, and there seems to be a resurgence in popularity with every generation, so it comes as no surprise that there are several emulations and 're-imaginings' now on the market. Strymon have a lot of experience in creating modulation effects, of course, including the notoriously difficult-to-model rotary speaker, and now they've turned their attention to a single-function pedal dedicated specifically to the Uni-Vibe sound.

The USA-built Ultraviolet is housed in a purple metal case with all the connectors on the top edge of the pedal. A 9V clean supply of 250mA minimum is required (the PSU is not included). As with the other pedals in the range, all three permutations of mono/stereo in/out routing are available via a rear-panel switch, there's a USB-C port for handling firmware updates or MIDI, and there's an Ext/MIDI jack, which can be set to one of five modes: Expression Pedal, Favourite, Tap Tempo, Infinite or MIDI. Stereo operation requires a TRS adaptor to split the feeds in and out of the pedal. Up to 300 presets can be stored and recalled using MIDI, though there's no front-panel preset feature.

The control set is a little different from the other two pedals we've seen so far in this series, insomuch as there are only three rotary controls and a pair of three-way toggle switches on the top panel. The three knobs set Speed, Intensity and Volume, while the left-most switch selects between Chorus, Blend and Vibrato. Chorus uses an even mix of the wet and dry signals, while Vibrato uses only the wet signal. Blend is a halfway house between the two. The other switch selects Low, Mid or High Bias and essentially sets the mid point of the LFO modulation. Low produces more pronounced low-frequency modulation while Mid and High move the effect further up the audio spectrum. I've often found the Uni-Vibe effect to sound a little 'lumpy' in comparison with a regular phaser and that sense of asymmetry is replicated very accurately by the Low Bias mode. For a less lumpy, generally brighter-sounding result, the other two Bias positions offer very usable alternatives.

Strymon's pedals always attract praise for their sound quality and the accuracy of their emulations, and the Ultraviolet is going to be no exception. It is perfectly capable of delivering an authentic Uni-Vibe sound that, at higher modulation rates, has an almost rotary speaker quality to it. However, the switchable Bias and Blend options genuinely extend its usefulness, and my personal preference when recreating those languorous slow sweeps (in Chorus mode) is to use the High Bias setting in combination with the Intensity control just past half way. Vibrato mode produces an organic-sounding pitch modulation while Blend is useful if you want a more subtle version of the chorus effect. Having those switchable options makes it easy to find your own Goldilocks setting — not too lumpy, not too smooth, but just right! I have a feeling that this pedal is going to do very well amongst those who really appreciate the Uni-Vibe sound. *Paul White*

$ $259.
W www.strymon.net

Warm Audio ODD Box v1

Overdrive Pedal

Texas-based Warm Audio established their reputation through building affordable replicas of mostly out-of-production recording-studio classics. Widening their portfolio to include guitar pedals saw the launch of the Foxy Tone Box (a rare vintage '70s fuzz pedal) their Centavo Klon Centaur 'clone' and the distinctly Zen-like Warmdrive. Their newest pedal, the ODD Box v1 (ODD standing for Over Drive Disorder), however, pays homage to one of the all-time pedalboard favourites, the Fulltone OCD pedal. Designer Mike Fuller's Fulltone business closed down its Culver City operation in September 2022, seemingly for good, but has since resumed production of some products from his new base in Nashville. The current portfolio does include an OCD, but only in the form of the significantly revised version 2 (now at 2.1), released in 2017 and incorporating active bypass circuitry.

The original OCD design went through a number of iterations, all still officially called version 1, perhaps in recognition of the fact that they were all just relatively minor revisions of the basic circuit. Pot values changed, capacitors came and went and LEDs changed colour, but they all still sounded like OCDs, to me at least, and I have nearly all of them, including the extra-rare v1.5. That didn't stop people having very clear favourites, though, with the v1.4 and v1.7 guaranteed to top any forum poll of 'best OCD version'.

What all the versions have in common

is the unusually high degree of touch sensitivity they afford the player, offering a level of distortion control just through picking strength that is quite unlike most distortion pedals and more like the feel of playing an overdriven tube amp. Instead of the clipping diodes connecting to ground, like most hard-clipping pedals, the OCD uses the input of the op-amp stage as a floating bias point, which means the clipping threshold ends up being proportional to the strength of the input signal. The OCD also employs MOSFET transistors for clipping, albeit used simply as diodes, and this too contributes to a slightly softer 'feel' than a 'silicon-diodes-to-ground' pedal.

Warm Audio's ODD Box replicates the circuit of the 1.4 OCD variant, interestingly one that exhibits a couple of things not present in any other version. The first of these is the presence of an additional germanium diode, and the other is the use of a logarithmic volume pot in place of the linear pot used in all the other v1.x models. This places the unity volume point much higher up on the rotation of the pot and led to the widespread misconception that the 1.4 was 'quieter' than other versions, largely based on YouTube videos comparing it to other variants with the controls in the same physical position. Of course, it isn't quieter, it just gives you a lot more control within the most useful bit of the range. The ODD Box has it, too, and I much prefer it to linear pots that cram everything up to unity in the bottom quarter of rotation.

The effect of the 'extra' capacitor is harder to discern because other components also change from one variant to the next — capacitors subtly change, pot values alter. In theory, the D4 diode should make the clipping more asymmetrical and more aggressive sounding, but my v1.3 sounds even more aggressive and biting, and overall I prefer the more balanced voicing of my 1.4, which I have to say I hear perfectly replicated in the ODD Box. The controls are a simple Volume, Drive and Tone setup, with a voicing switch offering 'UK' and 'USA' in place of the OCD's HP (High-peak) and LP (Low-peak). The legending may be different but the audible effect is precisely the same, with 'UK' High-peak exhibiting a Marshall-like bark in the upper midrange, compared to the flatter response of the Low-peak.

Inside, the Warm Audio pedal looks very tidily put together, with quality jacks and a three-pole switch offering full mechanical bypass as well as LED switching. Disassembly for battery access is via side-located thumbwheels, as found on v2 OCDs, which is certainly convenient when you don't have a screwdriver handy.

No 'version 1' OCDs of any variety are currently in production and the secondhand price of the favourites like the 1.4 and 1.7 have risen accordingly. To some people there will always be an extra kudos in having the 'real thing', but if you want a v1.4 and can't justify the inflated pricing, Warm Audio's ODD Box will get you there. The launch of this pedal set the guitar forums on fire with the usual quasi-legal speculation, but just to be clear, you can't patent a circuit, and this does not infringe Fulltone's 'trade dress' (the relevant trademarking principle here): it is a different colour and it clearly says Warm Audio on it. Nobody is ever going to mistake an ODD Box for a Fulltone OCD by looking at it. Now, a blind listening test, that's a different matter... *Dave Lockwood*

$ $119.

W www.warmaudio.com

SUB ZERO

Negative Ratio Compression

Turn the ratio up, and a compressor becomes a limiter. Go even further and you enter the world of reverse dynamics — and unique sonic effects.

MIKE SENIOR

If you're reading *Sound On Sound* then I imagine you're already using compression regularly, so you'll know that the traditional range of compression ratios stretches from 1:1 (in other words, no compression at all) to infinity:1 (hard limiting). However, even at infinity:1, signal peaks are only limited to the level of the compression threshold — and there are some situations where this can be, if you'll pardon the pun, a limitation!

For example, let's say you've triggered a snare-drum sample alongside your live drum kit recording, but you only want to add the sustain phase of the snare sample, not its attack transient or the body of the hit. You could try to use compression to turn down the level of the sample's onset relative to its sustain, but no matter how high you set the compression ratio, the sample's sustain is never going to end up any louder than its attack. Even limiting will just put the attack and sustain components at the same level.

Beyond The Limits

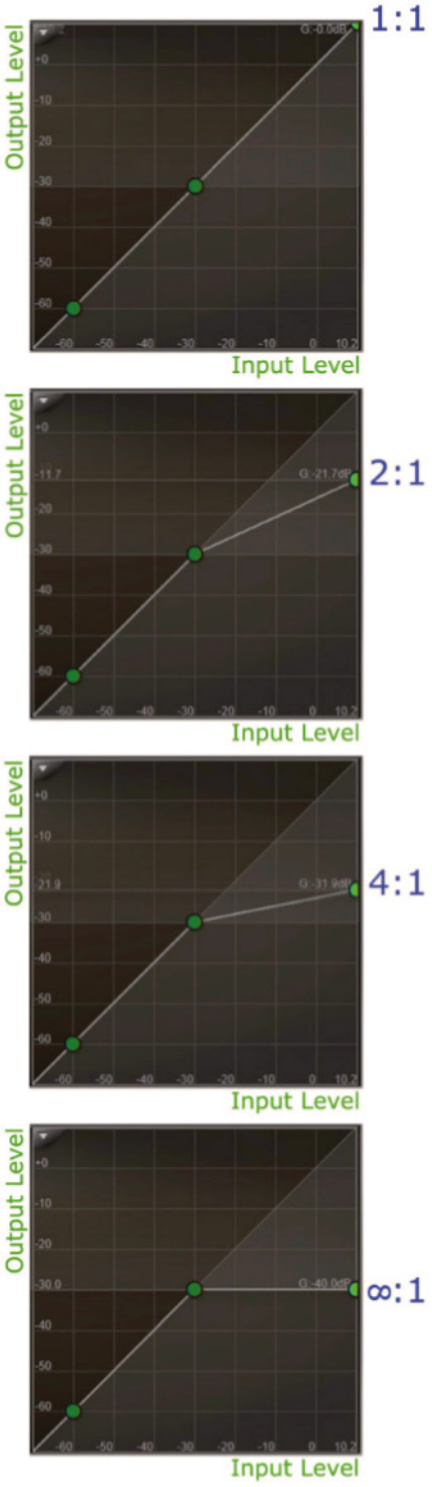
As it happens, though, there are a few unusual compressors that offer a solution to this problem by offering negative compression-ratio settings. To explain how negative compression ratios work, imagine an analogue signal level that measures +12dBu feeding a compressor with its threshold set at 0dBu. At a ratio of 2:1 the signal would be exceeding the threshold by 12dB, so the compressor would apply gain reduction to give an output level only half as far over the compression threshold — in this case 6dB over 0dBu, which means an output level of +6dBu. On the other hand, if you changed the compression ratio to a negative setting of -2:1, the compressor's output level would end up being 6dB below the threshold level, rather than 6dB above it — in other words, 6dB under 0dBu, which would mean an output level of -6dBu.

So, if we now return to our previous snare sample scenario, let's say each hit's peak level is +12dBu, and you compressed it at a threshold level of -12dBu, even the highest traditional compression ratio of infinity:1 would only knock the drum's attack down to -12dBu. With a negative ratio of -2:1, however, the attack would be turned down to -36dBu, 24dB below the level of the sustain phase — which would remain uncompressed, because it's below the threshold. With the snare sample's attack now so far below its sustain in level, mixing it in alongside your live drums would enhance the combined snare sustain much more independently.

Now, to be fair, there are other ways to strip the attack off a simple triggered sample — adjusting the ADSR envelope settings in your sampler, for instance, or applying a slow-attack gate to the audio. But where I think negative-ratio compression really comes into its own is as a means of enhancing the sustain of live snare-drum recordings directly. Here's what to do.

Processing Live Drums

Firstly, set up your negative-ratio compressor as a parallel processor — in its own return channel, in much the same way you'd configure a reverb or delay send effect. Next, send from your snare-drum close mics to the parallel compressor and adjust the threshold and ratio settings to all but silence the snare peaks. (A negative ratio of -1:1 will usually do the trick as long as you set the threshold low enough.) Then mix

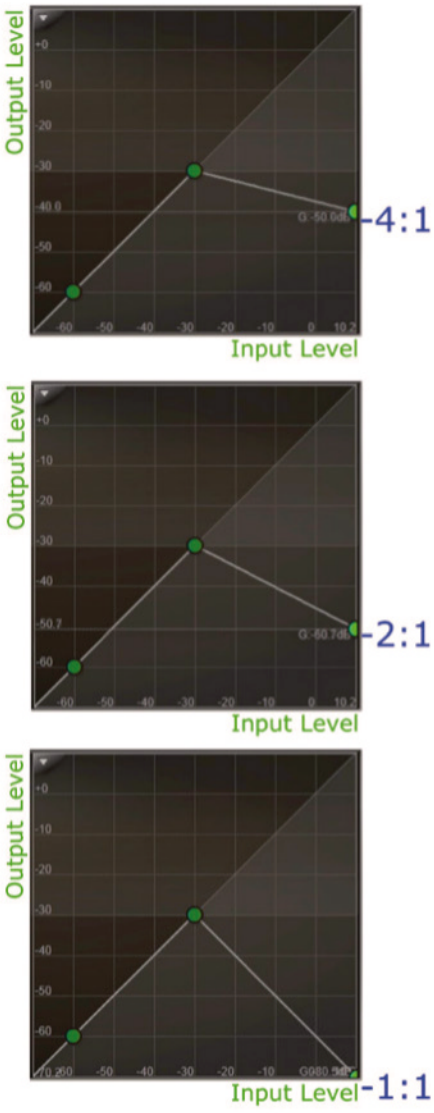


The four diagrams above show transfer functions for the common compression ratios 1:1, 2:1, 4:1 and for traditional infinity:1-ratio limiting. The three diagrams on the right illustrate the much less common negative compression ratios of -4:1, -2:1 and -1:1.

the compressed signal back in with the uncompressed live drum tracks to boost the snare sustain.

Now, you might well ask why I wouldn't just use normal channel-insert compressors instead. After all, if I just compressed the snare peaks and then applied some make-up gain, wouldn't that also boost the level of the snare sustain relative to its attack? Yes, it would... But unfortunately that would also iron out the player's performance dynamics, and might soften the drum's attack transients, too. Another advantage of having the sustain enhancement coming from a separate mix channel is that you can also reshape the tone of the added sustain by just EQ'ing the negative-ratio compressor's return channel.

Why not just use standard parallel compression on the snare close mics, though? Well, the problem here is that mixing that channel in with the uncompressed live drum kit tracks wouldn't just boost the snare sustain — it'd also boost the snare's attack/body to an extent too, and therefore the snare's peak level in the mix. With negative-ratio compression, you can pretty much silence the snare peaks in the parallel signal, which means you can freely mix in the compressed channel to enhance your snare's sustain without appreciably changing the drum's peak level in the mix at all. As such, negative-ratio parallel compression won't upset the action of any bus compression you might be using further downstream in your mix setup, and the snare sound also won't be affected as much if your final mix's peaks end up being steamrollered by loudness processing at the mastering stage.



A Question Of Timing

Once you start applying any serious amount of negative-ratio compression, you quickly discover that your compressor's time response is crucial. If you're trying to stamp on a drum transient, and the compressor can't clamp down quickly enough, a spitty little sliver of transient can easily still make it past the compression, which usually isn't ideal for the mixing applications I talk about in this article. But even a very fast-acting compressor may not be able to apply enough gain reduction quickly enough to completely remove all trace of a drum transient, and if that's what you're looking to do, then you should seek out a compressor with a lookahead function. That way, you can give the compressor's gain reduction time to kick in fully a few milliseconds before the transient actually arrives. Alternatively, if your compressor has no lookahead parameter, you can still implement lookahead functionality so long as the plug-in has an external side-chain input, by feeding a pre-delayed trigger signal to that.

Layering Loops & Breaks

Another situation where negative-ratio compression really comes into its own is when you're mixing multiple drum loops together, or where you're layering a breakbeat or two over a basic programmed beat. Both these techniques are very common in pop, electronica and hip-hop production, because they're a quick way of adding textural interest to your rhythm parts — and if the loops or breaks in question feature elements of live drumming, they'll often help the programmed arrangements feel less mechanical too. One difficulty that occasionally arises with this tactic, though, is that timing variations between the different loops (or between your programmed beat and the break you've layered over it) undermine the overall flow of the rhythmic groove. They may even cause distracting flamming. And negative-ratio compression can be really helpful in such situations.

You see, I've found that there'll almost always be one main track that really defines the groove in most projects. It'll often be the programmed beat rather than the layered breakbeat, whereas with layered loops, it'll typically be the loop with the most powerful kick and snare hits in it. So what I'll do is identify which loop or break is conflicting with the main rhythmic track, and then use fast negative-ratio compression to aggressively gain-reduce its peaks. You might think this would defeat the object of the layering, but in practice I think the reason most people add extra loop/break layers to their arrangements isn't because they want to enhance the transient peaks — it's because the additional layers add texture, depth, width and general organic character. On this basis there's usually very little down side, in my experience, to using negative-ratio compression to resolve groove conflicts by aggressively removing the peaks of subsidiary layers.

Extreme Pumping

A final cool application of negative-ratio compression is for implementing the kinds of extreme level-pumping effects you hear on some EDM records. Clearly, if you're just wanting to pump the level of a trance pad with a four-to-the-floor kick drum, then there are already lots of ways to do this, such as tempo-sync'ed gating or LFO amp-modulation — or even just a simple sawtooth automation-data pattern. However, the moment you want to trigger those kinds of pumping effects using more unorthodox kick-drum patterns, or with multitrack recordings that were recorded live without a click, the options become more limited and/or labour-intensive.

You could use a conventional compressor or limiter (triggered through its side-chain by your kick drum) to implement a pumping effect in such cases, but neither will give such a strong effect as a negative-ratio »

» compressor, which can pump right down to silence if need be. You might also try a side-chain-triggered ducker instead, but although that might be able to offer a similar severity of gain reduction, its pumping effect isn't capable of tracking any dynamic variation in your kick-drum trigger part like negative-ratio compression can — duckers are basically just on/off switches.

Now, clearly negative-ratio compression is a niche mixing tool, and I'm not saying that it'll always be the best tool for every task. But I'm finding myself turning to it more and more these days for its unique problem-solving properties, and I hope this little article has inspired you to experiment with it as well! ■■■

Some Compressor Plug-ins That Offer Negative Ratios

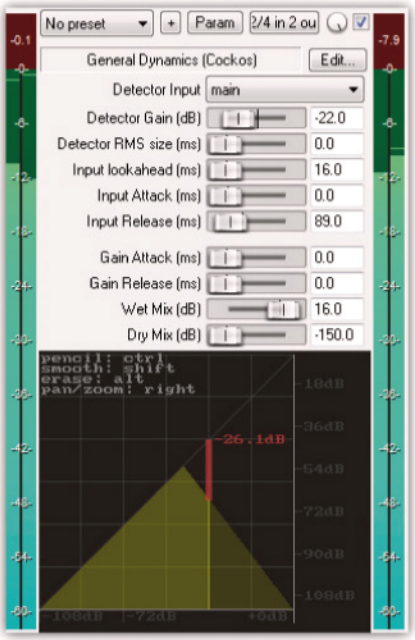
The classic negative-ratio compressor is the **Eventide Omnipressor**. Long since discontinued in hardware, it lives on in plug-in form:

W www.eventideaudio.com/plug-ins/omnipressor

There are also quite a few free options.

Cockos ReaJS General Dynamics: This lets you draw any transfer function you want including negative ratios, and also includes lookahead. The Mac version only works in Reaper, though.

W https://www.reaper.fm/reaplugs



Klanghelm DC1A: Cross-platform, lots of fun and super easy to use, this freeware gem offers negative-ratio compression as a switch option. However, the control set is very small, there's no lookahead, and some of your drum's attack still seems to break through even at the most extreme settings, so I'd not recommend it for serious negative-ratio mixing duties.

W https://klanghelm.com/contents/products/DC1A.php

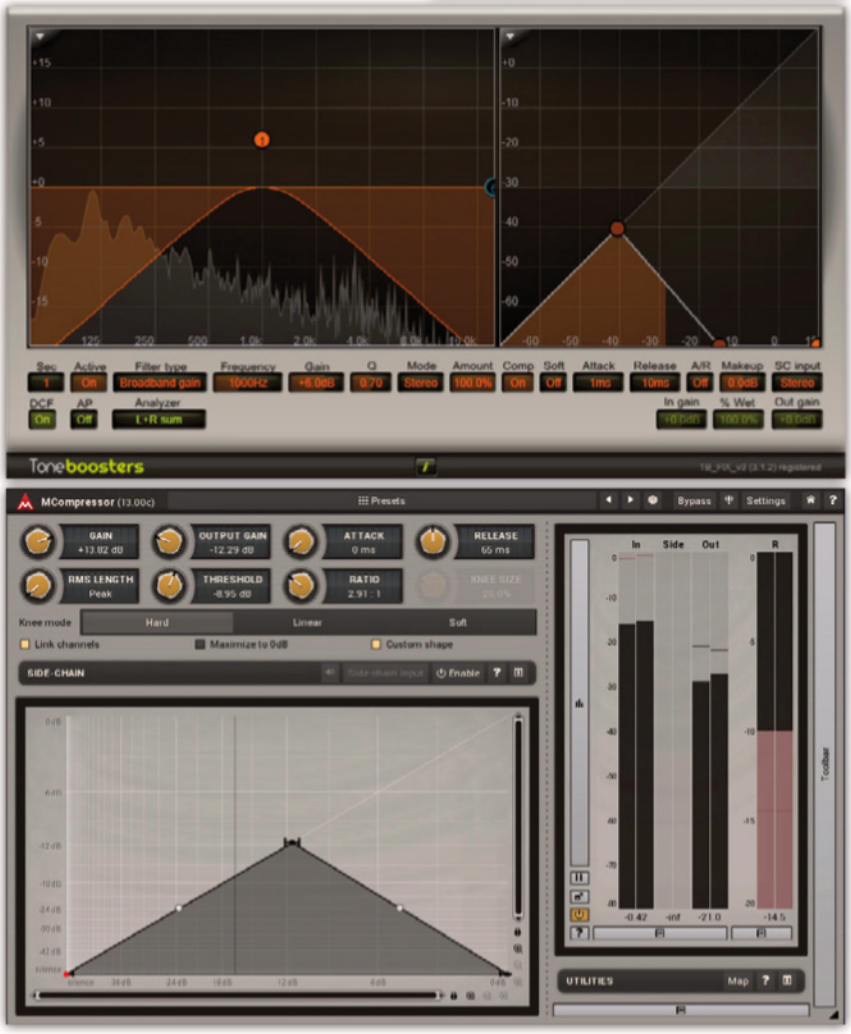
Melda MCompressor: Part of Melda's cross-platform MFreeFXBundle freeware pack, MCompressor will let you draw in a negative-ratio transfer function if you tick the little Custom Shape box. There are also lots of useful compression parameters to play with, but no lookahead, unlike some of their payware plug-ins such as MDynamics.

W www.meldaproduction.com/MCompressor

Toneboosters TB_FIX: When Toneboosters officially discontinued this former payware plug-in a couple of years back, they generously decided made it available as freeware on their website FAQ page under "Where can I find older plug-ins?" (it's part of the BusTools bundle).

Although TB_FIX is primarily a dynamic EQ (and still my own personal favourite!), there's a Broadband Gain filter type you can select for any of the EQ bands, which effectively turns it into a standard compressor. You can then draw in a negative-ratio transfer function for that band. There's even a side-chain input if you want to manually implement a lookahead facility.

W www.toneboosters.com/support.html



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Calibrating Your Mixing Setup

Mixing gets so much easier when your processing chains are configured to deliver the same results in response to the same levels!

Recalling hybrid mixes is much quicker if you never have to adjust parameters on your hardware outboard.

Gain staging and reference levels have been hot topics in audio for years. But there's another, closely related and very important subject that I think gets overlooked: calibration.

At some point, and long before I used hardware gear myself, I understood that it makes sense to use a test tone when you want to recall an analogue mix or just to ensure that the output of an analogue device is the same as it was yesterday (believe me, that's not a given). But the idea that calibration using tones can actually be a creative tool itself didn't occur to me until I dove into Michael Brauer's 'Brauerize' technique about six years ago. Vance Powell has likewise shared valuable workflow lessons concerning calibration.

What's more, calibration is equally beneficial when working 100 percent in the box. There's a reason why Brauer and his assistant went the extra mile to reproduce the analogue Brauerize calibration settings as closely as possible when he went all-digital.

Calibration can do many things for you that are all very helpful. These are the most important ones:

- It will establish a familiar level structure that's the same for every mix.
- As a consequence, your mixes will be very consistent level-wise.
- It removes a lot of technical fiddling from the mix process so you can focus on creativity.

Let me give you a practical example from the studio I work in regularly. There's lots of outboard in the racks, all hooked up to the converters and set up as hardware inserts in the DAW. Because different people use the studio to mix, I proposed establishing a fixed calibration so recalls are super easy. First, compressor parameters such as attack, release and ratio were set so as to give a great sound as a starting point — and no-one has since felt the need to change anything. Our reference level is -18dBFS for +4dBU, meaning a test tone at -18dBFS within the DAW measures 0VU on an analogue meter. So I sent

a -18dBFS test tone from the DAW and calibrated every compressor so that this tone generated 3dB of gain reduction, with 3dB make-up gain applied on the output, so that they all return a -18dBFS signal back to the DAW. This means that when fed with a signal conforming to the gain staging and level structure we tend to work with, they give a predictable amount of compression while maintaining a consistent level. This plays nicely with the way hardware inserts work in most DAWs: if you want more or less compression, you can simply send more or less signal from the DAW, meaning that as long as no-one changes the parameters on the hardware compressor, saving and loading a DAW session is enough to fully recall the mix.

All these calibrated devices can also give you valuable information when your gain staging is off. If a lightly picked acoustic guitar yields 6dB of compression right off the bat, it's an indicator that it may be unusually loud. Of course that may be exactly what you want, but it's useful to have that warning.

Going Soft

Calibration is equally useful if you mix entirely in the box. I suggest taking your favourite compressor plug-ins and making presets that do exactly what I described above. Create a tone at your reference level, be it -18, -20, -16 dBFS or whatever you use, adjust the compressor parameters to a setting you typically use or start with, set the threshold to deliver 3dB of compression (or 1dB if you prefer) and the output level to maintain »

Speaker Calibration

As well as gain staging, the word 'calibration' is often used in association with studio monitoring. Many people consider it good practice to have a monitoring setup that is configured to give a fixed sound pressure level in response to a known input level. Like calibrating your processing chains, this can make it easy to hear straight away whether things are too loud, too quiet, too bright or too dark. But calibration can also mean using a tool like Sonarworks Reference, Trinnov or Genelec GLM to correct the frequency response of your speakers — and in an immersive setup, it also involves time-aligning the many speakers to optimise time of arrival at the listening point.

For more detail on calibrating monitor levels for mixing, read Hugh Robjohns' article from *SOS* May 2014: www.soundonsound.com/techniques/establishing-project-studio-reference-monitoring-levels

How To Calibrate A Plug-in Compressor

First, mute your speakers! You need a plug-in generating a sine-wave test tone at your chosen frequency and level. Every DAW should have one, and if yours doesn't, there are free plug-ins out there. For general calibration, I recommend the standard 1kHz sine wave. You could use your DAW's meters, but I recommend using a VU meter, which many DAWs also have on board. This has the advantage that your DAW's pan law will not affect the metering. There's free VU meters, or Klanghelm's affordable and excellent VUMT. Now create a mono and a stereo track in your DAW, insert the test tone generator in the first slot of each and set it to sine wave, 1kHz and your chosen reference level. As said, I'm using the popular standard of -18dBFS. Now insert the VU meter on the third slot and verify it reads 0VU. If not, check the VU's internal calibration.

Now insert the compressor to be calibrated in the second slot (use a mono instance on the mono track, stereo on the stereo track) and set parameters such as ratio, attack and release to your most-used settings. Then, depending on the topology, adjust the threshold, input or peak reduction control until the compressor shows you the desired amount of gain reduction (I recommend no more than 3dB). Now adjust the output or make-up gain until your VU meter reads 0VU again. Save either as default or as a preset. You can also save presets for 1, 2 and 3 dB of gain reduction, or presets using different ratios. Usually, the lower the ratio, the more gain reduction you can start with.

Calibrating a UAD 1176LN compressor plug-in in Cubase. The compressor is instantiated in the second insert slot in the channel, between a tone generator and a VU meter.

» unity gain. You can even save this as the default preset if you wish.

Where I find calibration even more important is on busses, because then you essentially mix 'into them'. For instance, in my personal hybrid home studio setup, I have all my multi-bus compressors calibrated to give the desired amount of compression and maintaining unity gain with a -18dBFS stereo tone. My final master bus compressor then is calibrated to give 1dB of gain reduction with a -14dBFS tone and unity gain. This works super well when using a -18dBFS reference, since all those signals add up

Calibration In The Real World

To illustrate the points made in the main text, it might be helpful to provide a quick rundown of my own hybrid setup and how it's calibrated. Here's how the most important parts are set up to work together, based on a calibration level of -18dBFS=0VU. My two stereo main multi-busses are calibrated as follows:

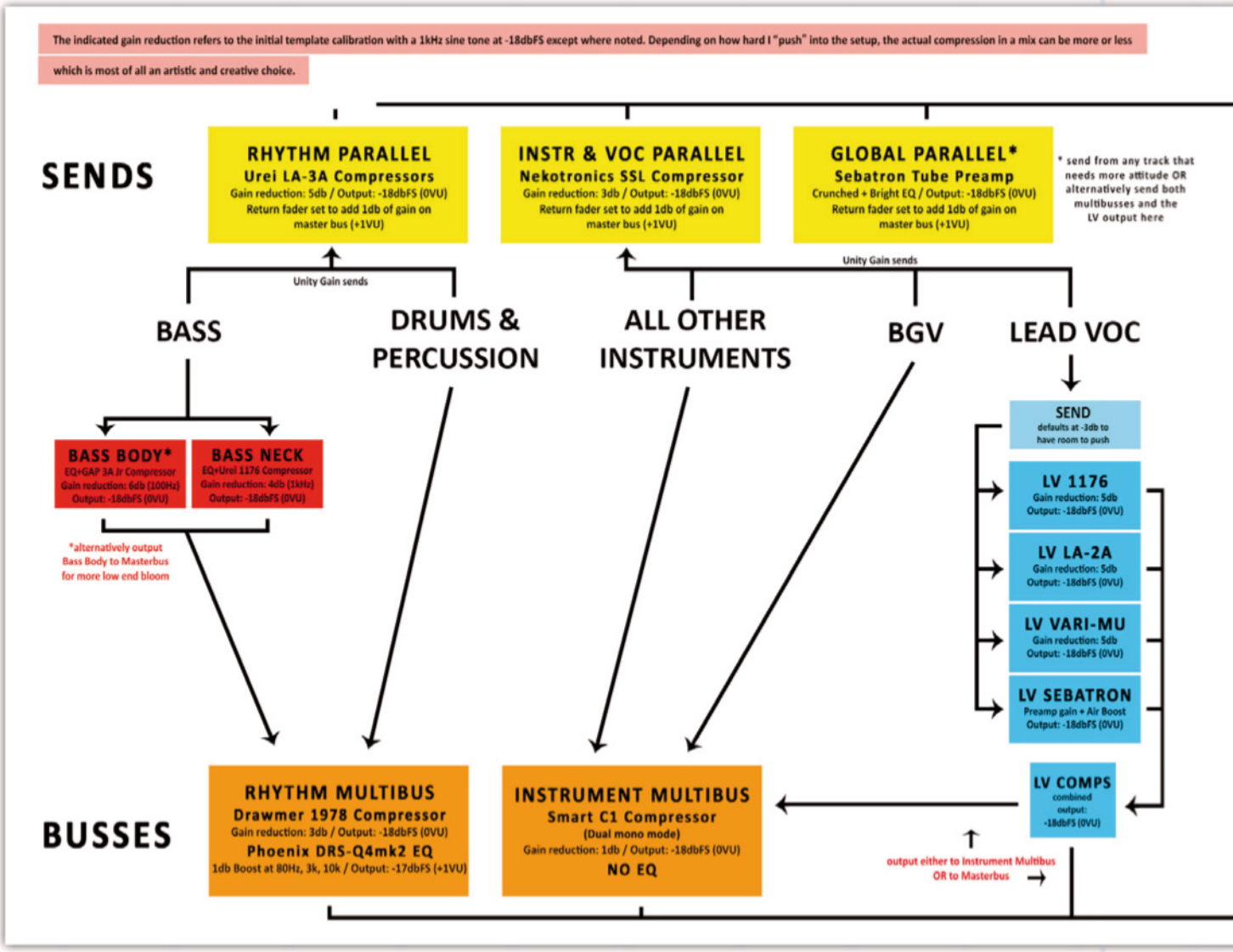
- Rhythm (drums, bass, percussion): 3dB of compression, unity gain, plus EQ boosting lows, mids and highs each 1dB with appropriate test tones, combined output +1dB.
- Instruments and effects: 1dB of compression, unity gain, no EQ.

My mono vocal busses are each calibrated to give 5dB of compression at unity gain (though I typically start the mix with the send lowered to give between 1 and 3 dB and leave room to push). My main parallel busses are calibrated as follows:

- Rhythm parallel: 5dB of compression, unity gain, fader set to add 1dB of level to the mix.
- Instruments and vocal parallel: 3dB of compression, unity gain, fader set to add 1dB of level to the mix (since so many signals are sent to this parallel, it usually compresses between 4 and 6 dB in practice).

My whole master bus chain is calibrated to -14dBFS=0VU (4dB over reference) and set up as follows:

- SSL Fusion: drive set to my liking, output compensated for unity gain.
- Insert (pre-EQ): SSL XR626 compressor set to give 1dB of gain reduction, unity gain.
- 1.5dB boost at 50Hz and 1dB boost at 12kHz, measured with corresponding frequencies on the tone generator.
- HF compressor set to bring the 1dB of high boost back to unity with the 12kHz test tone.
- Stereo image to my liking, transformer on.
- Output further adjusted for unity gain with the whole chain.



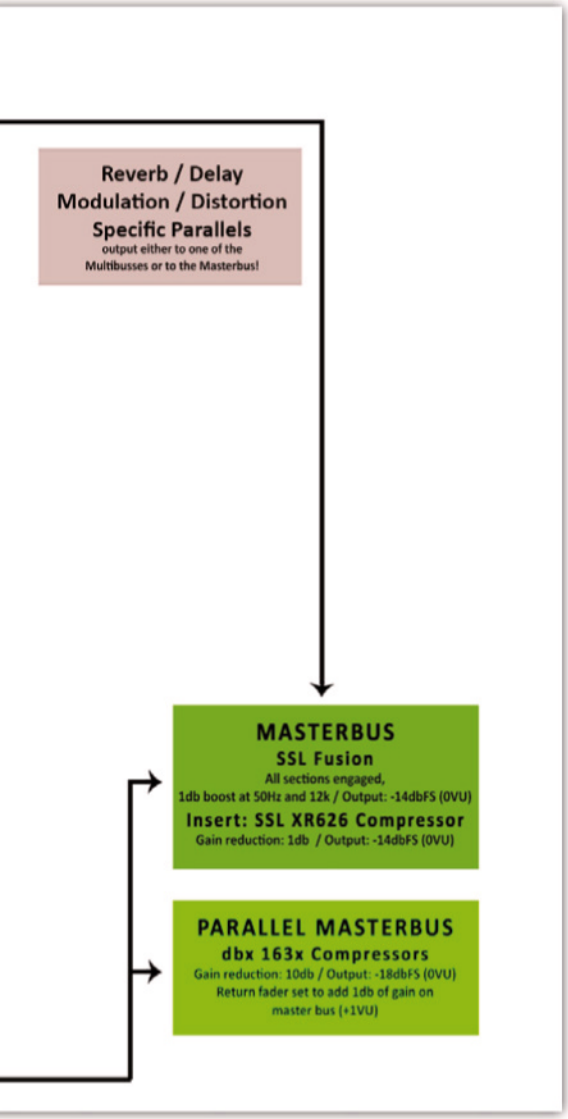
at the master bus. I have the compressor on from the beginning, and when I hit it too hard, I know the mix is too loud. Why would I adjust the compressor's threshold every time to suit the mix, when I can make the mix suit the threshold?

You can even calibrate your EQs. Let's say you use some kind of Pultec on the master bus, which is very popular. If you want a familiar starting point, set the tone at your preferred master bus level (as stated, that's -14dBFS in my case, meaning 4dB over my reference), switch the tone frequency to 100Hz and note the level your DAW meter is showing. Now turn up the low frequency of your choice on the Pultec to add 1dB of level. Switch the frequency on the tone to 10 or 12 kHz and turn up the EQ's high frequency of your choice to add 1dB of level, too. (You'll realise those increments on a Pultec are not accurately labelled in dB!)

But it doesn't stop there. A beautiful way of calibrating your parallel processing chains is to follow Vance Powell's method. Instantiate a test tone of your reference level on an audio track and route it to the master. Now also send it at unity gain to your parallel chains one at a time and set the latter's return faders so the level on your master bus goes up by 1dB. Depending on the nature of the parallel chain, test tone frequencies other than 1kHz may work best; Vance Powell likes to use 400Hz because he feels that's more "in the music" than the standard 1kHz tone. For parallel bass or kick drum processing, 100Hz may be the way to go.

I find this method to be a great starting point for all kinds of parallel processing, but feel free to adjust it to your needs. Oh, and of course it would be good practice to calibrate the parallel chain itself beforehand. If you want your parallel drum compressor to give about 6dB of compression, use your reference tone to set it up. In case you use a tried and true setting on a parallel that just works, leave it be. Since we are adjusting the fader anyway, unity gain is less important within a parallel chain.

The author's calibrated hybrid mixing setup.



Speed Boost

After years of using and fine-tuning my calibrated mixing setup I have got it to a point where just by routing and sending the tracks to the busses and parallel chains in my template, everything falls into place immediately. I start by finding my initial balance, just using the faders and pan controls, with all tracks routed to the final stereo bus and the final master bus compressor engaged so I get a feeling of how loud the mix can and should be. By now I know exactly where the drums should be level-wise, so I typically start with those and adjust everything else accordingly. When I'm happy with the balance and the response of the master bus compressor, I export this state as a reference and then route and send everything to the calibrated multi-busses and parallels. Depending on the track count, this takes two minutes at max, and when I hit Play and compare against the reference state, everything immediately sounds 80 percent more like a record: more glue, more punch, more excitement, just like that. If later in the mix I should feel I'd better back down, I have a master VCA in place to turn down all tracks whilst maintaining their relative balance. This way, I can easily adjust how hard I want the whole mix to 'push' into the setup. Sometimes I end up where I was, sometimes I end up as much as 3dB lower; this is just about listening and knowing when I've hit the setup's sweet spot.

Some final words: this should only serve as an appetiser in case you haven't yet made use of this powerful tool, or a reminder of how beneficial it can be for your workflow in so many different places. In any case, make sure you find a calibration that works for you and you alone, and stick to it. If, after doing a couple of mixes, you feel the same things don't really fit, adjust accordingly. Once you've found your own calibration, you'll be able mix with confidence and speed. Who wouldn't want that?

Parallel Sends: Unity Or Bottom-up?

Generally, there's two approaches for using sends: you can either start way down and raise the sends until you hear the desired effect, or you can leave the sends at unity gain and only adjust the return levels. Both approaches work with pre- and post-fader sends, and if you use the second approach with post-fader sends, what you get is an exact copy of your fader balance. This approach works really well for parallel processing that is shared across many tracks, Andrew Scheps' infamous 'rear bus' being a prime example. This also makes calibrating those parallels rather easy when using Vance Powell's method mentioned in the article:

Keep your speakers muted! Set up your first parallel chain and send to it from a tone generator track at unity gain. Pull the parallel chain's return fader down. This time, insert the VU on the master bus. With your active tone generator track routed to the master bus, it should read 0VU for a stereo tone or a lower value for a mono tone, depending on your pan law (most likely -3dB). Now raise your parallel's return fader until the VU on the master bus has gone up by 1dB. Repeat this process one at a time for every parallel. Note the fader values in the template for later reference. This way, you make sure your most important parallel processing all starts adding the same level to the mix. I do ride my two main parallel busses on almost every mix, but I always start at the default values just because it works so well and the overall mix balance isn't changed. Feel free to experiment with adding different gain values and find out what works best for you.

THE WINDT HUMMER TEST

Testing For Equipment Grounding Problems

If you want to identify studio equipment that's susceptible to grounding problems, a simple DIY project based on a wall-wart PSU can help.

HUGH ROBJOHNS

The author's DIY Hummer, made from an AC/AC wall-wart PSU.

My Ground Control article in *SOS* August 2023 (https://sosm.ag/ground-control) raised the topic of good and bad audio equipment design practices in respect of correctly managing interference and noise currents. In poor designs, unwanted noise currents from ground loops or RFI (radio-frequency interference) introduced via the shields of cables connected to other equipment are allowed to intermingle with the audio signal currents within the amplifier circuitry, adding hums and buzzes to the wanted audio. In contrast, good designs ensure such noise currents are kept well away from the audio circuitry, leaving the device blissfully quiet and trouble-free.

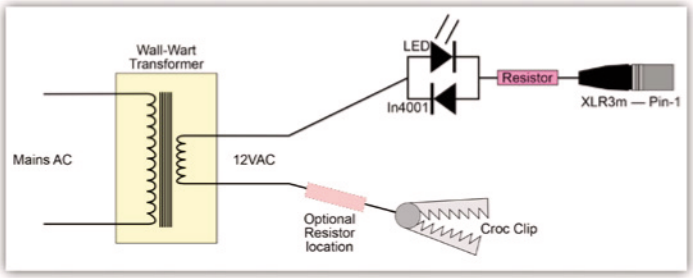
The Hummer

Unfortunately, there's no way of telling whether a piece of equipment is designed well or badly in this respect just by looking from the outside — but there is a simple engineering test to find out, and the necessary equipment can be constructed with moderate electrical DIY skills and a spare wall-wart power supply. This device, called a Hummer, was invented by John Windt and first described in the *AES Journal* (June 1995, Volume 43, No. 6). It was created in response to another article published in the same issue by Neil Muncy, who discussed at length equipment susceptibility to noise currents injected via external audio cables — something he christened the 'pin-1 problem'.

Conceptually, the Windt Hummer is nothing more than a reliable source of an audible noise current that can be injected (harmlessly) into the ground paths of any audio device. If a hum or buzz becomes audible (or the noise floor changes noticeably) at the output of the device under test, it clearly has a 'pin-1 problem' and will be highly susceptible to ground-loop or RFI issues when installed in an audio system. If all remains quiet under test, it is a well-designed device that handles noise currents properly. It really is that simple!

So, the source of noise current in the Windt Hummer is a standard (double-insulated) wall-wart power supply with an AC output. Any AC PSU with an output current rating of more than 30mA and an AC output voltage of between 6 and 24 Volts is acceptable — my own tester was built using a redundant 1.9A 9V AC PSU from an old phone charger that I found in the back of a drawer. The wall-wart transformer provides a safe, electrically isolated (ie. floating) and low-AC signal at 50 or 60 Hz (depending on the local mains supply). The actual voltage isn't particularly important, although I'd recommend 6-12 V if possible. The current being injected into the device under test is restricted using a current-limiting resistor. Windt's original design passed 100mA and that seems excessive to me, so the incarnation described here provides around 50mA of current.

Hearing a pure 50Hz hum might not be very easy in some listening conditions; a signal containing higher harmonics would be far more practical. Of course, the local mains supply may already have plenty of harmonic distortion, but a couple of diodes across the output to rectify the AC signal absolutely guarantees it! Moreover, if one of



Adapting an AC PSU to create a Windt Hummer test device is a very simple DIY job.

those diodes is an LED it will provide helpful visual confirmation that current is really flowing into the device under test.

The Hummer is designed to inject its noise current via an audio cable's shield connection, returning via the device's chassis (and/or the protective earth in the mains supply). A croc-clip can be attached temporarily to the chassis (or just held against it), so that's easy. Windt's original design used another croc-clip to access the audio connector shield connection (XLR pin 1) too, but I think that presents too great a risk of accidentally touching an audio input terminal (pins 2 and 3): a 12V AC signal is equivalent to +24dBu which, if accidentally touched to an audio input terminal, will max out a line input and probably destroy a mic input! So, to negate the risk of sending the Hummer signal anywhere other than the ground path, I *strongly* recommend using an XLR plug as the second connector, with the Hummer signal wired to pin 1 and nowhere else. If the device under test doesn't have XLR inputs, you can substitute a TRS plug (Hummer signal to the sleeve terminal) or stick with the XLR and use standard adaptor plugs or cables to convert from the XLR to any other format — that's what I do.

So, the complete parts list is: low-voltage AC wall-wart, red LED (ensure it has a peak current rating of more than 50mA), IN4001 diode, resistor (value from table shown), croc-clip, male XLR-3 connector, and heat-shrink sleeving. You'll also need a soldering iron and the ability not to burn your fingers or set the house on fire during the construction!

Depending on the construction of the redundant wall-wart power supply, it might be possible to build some of the components into the wall-wart itself, which I did. Alternatively, it might be more practical to build them into the XLR plug, with a small hole drilled into the side of the shell to accommodate the LED (with copious heat-shrink sleeving to prevent short-circuits in the wiring).

The current-limiting resistor's value and power rating depend on the output voltage of the wall-wart power supply. If you like sums and algebra, the calculations are from Ohm's Law in the form: R=V/I. We need 50mA (0.05A) of current, so with a 12V AC wall-wart the calculation is 12/0.05 = 240Ω. In practice, some of the AC voltage will be lost across the diodes, which means the actual current will be slightly less than 50mA, but that will help with power dissipation.

Talking of which, 50mA passing through the current-limiting resistor will make it warm, so it needs to be rated to handle that power dissipation. The formula required to check this is: Power = Voltage squared, divided by the resistance. In this example, that's 144/240=0.6W, and a 1W resistor will do the required job — it's not as if this Hummer is going to be left connected all day long, as current is normally only passing briefly during testing. If you can't find a calculator, the table below gives the required E24-series resistor values (5% tolerance is fine) and suitable power ratings for common wall-wart AC output voltages.

Wall-wart AC Voltage	Current-limiting Resistor Value (Ω)	Resistor Power Rating (W)
6	120	0.5
9	180	1
12	240	1
18	360	2
24	510	2

As I mentioned earlier, given a choice I'd recommend using a 6, 9, or 12 V wall-wart, since the current-limiting resistor will be physically smaller and easier to install inside the wall-wart or XLR. If there isn't space there, you can fit it to the croc-clip instead, supported and protected by some heat-shrink sleeving.

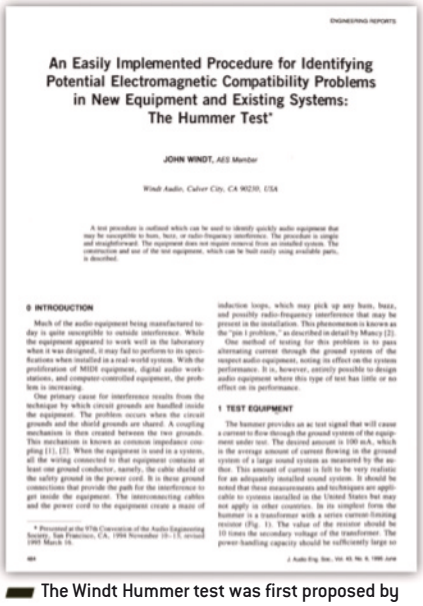
Testing

To test an audio device, disconnect all inputs and outputs except for the output being used to monitor for nasty noises. Turn the monitoring volume right down, then power up the unit and advance the volume control, listening for any hums or buzzes (there shouldn't be any at this stage!). If possible, advance the volume sufficiently to assess the level and nature of the noise floor (which is hopefully just normal Gaussian hiss). Make a note of this volume setting for later before turning it right down again.

Next, plug the Hummer's XLR into one of the unit's audio inputs and attach the croc-clip onto a suitable part of the chassis. The Hummer's LED should illuminate to indicate a noise current is being injected. If it doesn't, try the croc-clip somewhere else such as a fixing screw or the shield terminal of another connector. With the LED illuminated, carefully advance the monitoring system's volume while listening for hums and buzzes. If a buzz or hum is evident disconnect the croc-clip (the LED will extinguish) to confirm it is genuinely coming from the Hummer's noise current.

If the equipment is well-designed there shouldn't be any buzz or hum at all, but it's also worth checking the noise floor to see if the noise current has degraded it. Set the volume control to the noted position from earlier and check the noise floor remains at the same level and sound character as previously. If possible, check all of the device's inputs and outputs (via suitable adaptors) as some may exhibit problems while others won't, depending on the internal construction and wiring.

If the tested unit exhibits hum and buzz problems when using the Windt Hummer, it is highly likely to suffer from ground-loop noises and/or RFI issues when installed in an audio system using standard audio cabling. As a result you will almost certainly need to employ one-end-only cable techniques (see the aforementioned Ground Control article) or transformer isolation boxes to sidestep such problems.

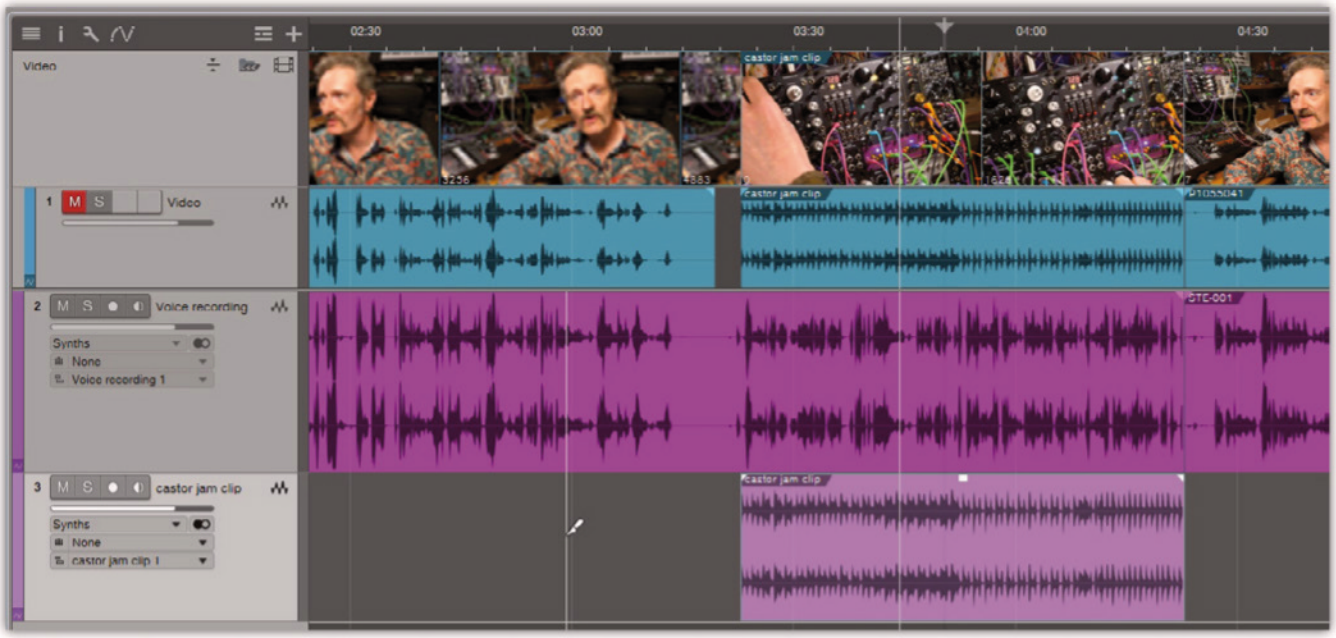


An Easily Implemented Procedure for Identifying Potential Electromagnetic Compatibility Problems in New Equipment and Existing Systems: The Hummer Test*

JOHN WINDT, AES Member

Windt Audio, Culver City, CA 90230, USA

The Windt Hummer test was first proposed by John Windt in an AES paper in 1995.



Dragging a video clip into Studio One will automatically create a Global Video Track.

Studio One is a surprisingly powerful video editor!

ROBIN VINCENT

In version 6 of Studio One, PreSonus revamped the Global Video Track from a simple player into something more useful. Video can now become an editable part of your project, letting you build podcasts, pull out TikToks, grab Insta reels or orchestrate your latest movie. I'm sure there are other possibilities between TikTok and eating popcorn at your local flicks, but that's a discussion for another time. The important thing is that working with film, footage, clips, video, grabs, or movies is now completely possible in Studio One. Let's check out how.

Elegant Or Simple?

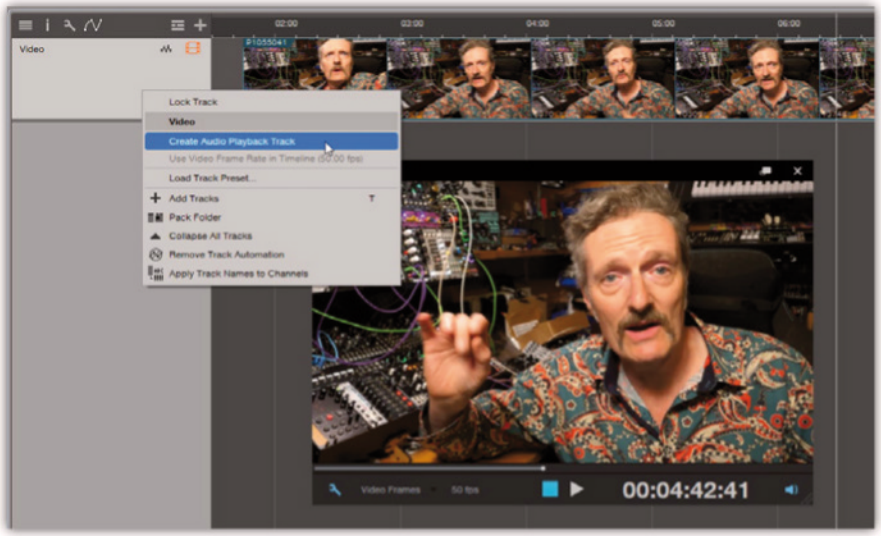
First of all, before we all get too excited, the video support in Studio One is rather elementary. This is not going to replace the depth and detail of Davinci Resolve or Final Cut Pro. There are just enough tools and possibilities to make it worthwhile exploring, but many of the vital things video editors need, like fades and crossfades, text overlay and colour correction, are unavailable. However, if you just shot a video podcast and need to take out all the coughs and "errs" then this is perfect. Alternatively, if you want to experiment with arranging video and audio clips side-by-side, this will do the job nicely. And if you are composing to video and need a bit of versatility and proper synchronisation when moving edits, then you will find what you need here.

The point is that a lot of the content we are creating is mixed media, and we often have to shift from one editing environment to another depending on what's driving our production. The Global Video Track doesn't solve that problem entirely, but it can keep you in the familiar space of Studio One more of the time.

In The Frame

As with most things in Studio One, the easiest way to import things is to drag them in from the browser. There's no limit to the number of videos you can pull in. Use the Files tab to find the folder with your video files; if they are compatible, they will appear with a filmstrip icon. In fact, I think drag-and-drop is the only way to import video files, and you can drag them in directly from a folder rather than using the built-in browser if you prefer.

The supported codecs will depend on your OS. Macs support QuickTime, MPEG-4 and M4V using the H.264 and HEVC codecs. QuickTime also supports MPEG and Apple ProRes. On Windows, you have MPEG-4 and M4V using H.264 and HEVC, and MPEG=PS using H.264 and MPEG. You will get the smoothest ride if all the clips within a single project use the same format and frame rate.



The Create Audio Playback Track option will put the video's audio onto a sub-track below the video.

Drag your video into any space, and Studio One will automatically create the Video Track at the top of the project in the Global Track lane. Alternatively, you can open the Video Track yourself by clicking on the Global Track Visibility icon and selecting it from the menu. It's the same place you'll find the Chord Track, Lyric Track, Markers and so on. Studio One generates thumbnails for you, and they zoom and scale really rather smartly. A quick tip is that if you hold Ctrl or Cmd when you drag in the video, Studio One will extract the audio and place it onto a sub-track for you.

If you are composing to the video then you probably want to be working in frames in order to keep the syncronisation as tight as possible. Studio One can detect the frame rate for you and set the timeline accordingly. There are a couple of places you enable this. First, you can right-click the video track and select "Use video frame rate in timeline", which also helpfully tells you what it is. Or you can get to it via the Song Setup window, where you'll find, next to the Frame Rate entry, a button that says "From video". Click that button! You can make all this happen automatically if you dig around in the advanced preferences under the Video tab and tick the appropriate box.

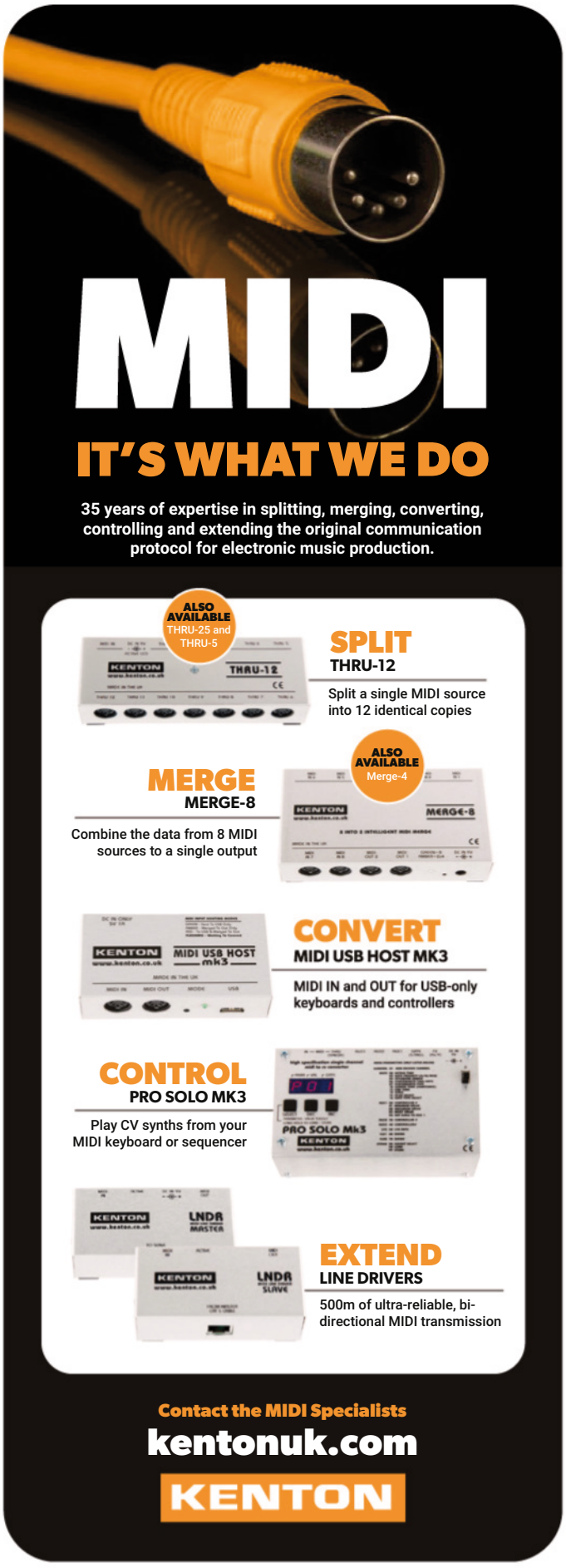
A few options are available in the inspector column if you press the 'i' above the video track. These include turning off the thumbnails and frame numbers, showing the audio track, and breaking the link between video and its audio for editing. Under a special Action button, you can set the project frame rate to the video, and remove the audio sub-track. You can also 'update' the audio track, although I'm not sure what this means. It doesn't appear to do anything, and there's no mention in the manual. Answers on a postcard, please.

Audio Playback

I mentioned how you can automatically extract audio when you import the video. The audio will exist whether you extract it or not, but by default, the Video Player has the audio muted. This audio might be the sound picked up by the camera microphone and may not be what you will use. In a podcasting scenario, for example, you would probably have recorded the dialogue on a better microphone, to a recorder or into Studio One directly to another track. So you can safely leave the Video Player audio muted. However, when editing the footage and the recorded audio, the sound picked up by the camera can be very useful in helping you visually align the separate audio to the footage. So, if you didn't automatically extract it, you can do so by right-clicking the video track header and selecting Create Audio Playback Track, or hitting the little waveform icon next to the filmstrip icon.

You can now line up your recorded audio with the audio from the video and mute the camera audio. You can also play fast and loose with the audio track because, after all, Studio One is all about the audio. The manual suggests that the sub-track is a bit basic when compared to a standard audio track but, to be honest, I can't find anything you can't do with it. It has its own channel in the mixer and access to all the editing tools, automation, effects and Event FX. But if you want to turn it into a normal audio track, all you have to do is drag it down to a new track.

Going the other way is perhaps more interesting. You can drag your recorded audio track up into the video sub-track and it will sit over the top of the original audio. The useful thing about that is that it will now move and can be edited with the »



video, as if it were the original audio. However, you can unlink the audio from the video using the little Link Audio Track Editing button that's appeared next to the waveform (which has turned into an open folder) and filmstrip icons. So essentially, provided you've lined them up correctly, you can replace the dialogue, then edit it with the video.

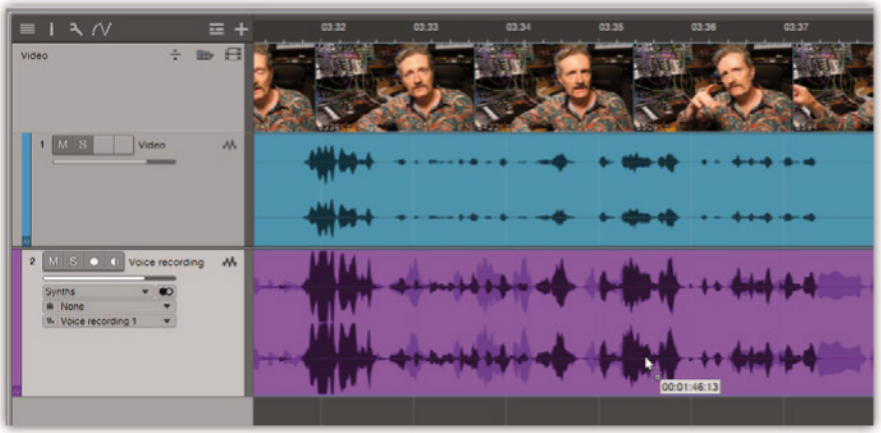
Edit Sweet

You'll be pleased to hear that the majority of the audio editing tools also work on video. This includes the trimming of the front and back edges, using the knife to split the video, cutting, copying and pasting, duplicate and slip editing (which is where you move the contents rather than the container). Ripple editing works, too: when enabled, this closes the gaps between two clips if you delete something in between. You can also use the Replace function to drop one video onto another and retain all the edits. Simply hold Alt or Option when dragging in the video. This could be a new version of the video, or a colour-corrected one, and so on.

One scenario I find myself in is where I want to cut a video clip of a product being used into the video of me talking about it. You can do this in the Video Track simply by dragging the product clip over the top. However, this will also replace the audio and is a reason, perhaps, why you wouldn't want to drop your recorded audio into the sub-track. In any case, you can pull the product audio out onto another track and drag the original audio back across the chunk that's been eaten away.

The video player window is excellent for scrubbing and fine-tuning your edits. If you press the little spanner icon, you can enable the Time Display. This is the time and frame display of the video clip only, and it doesn't necessarily match the project timeline. If you place the mouse over the time display and roll your mouse wheel, you can scrub the video in very precise frames. You can also do this on the main transport bar. I'm very impressed with how fast and smooth this is. It doesn't, however, scrub the audio track. You have to be in playback for that, which just ends up in a fight between you and the cursor.

Here's one tip if you are in a scenario where you are choosing takes or compiling short clips from longer pieces of video. Open up the Arranger track and use it to name all the sections of your edit.



Even if you don't use the video's original audio, it can still be useful when it comes up to lining up your final audio.

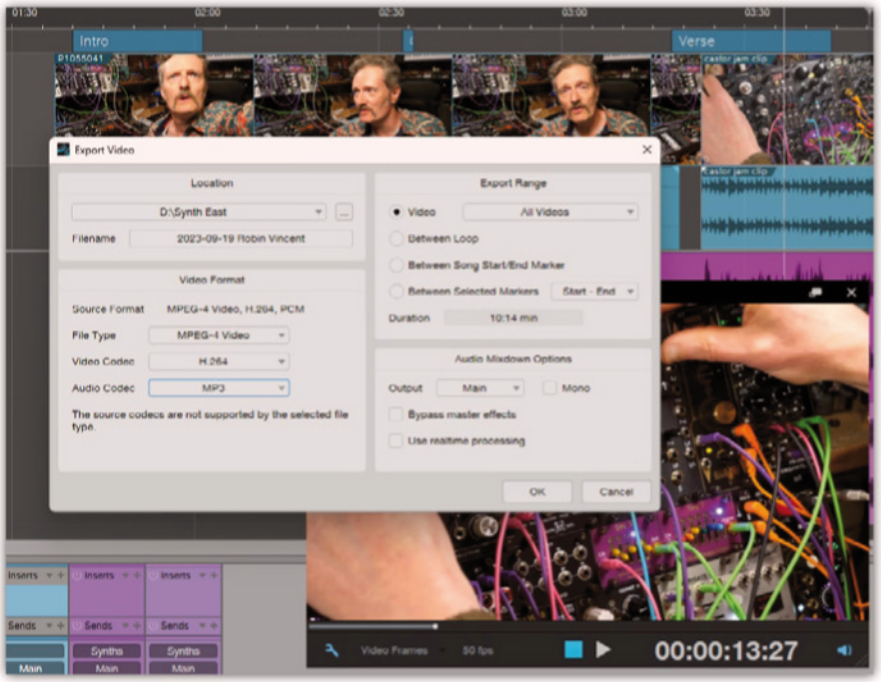
When you're ready to choose the clips, simply open up the Arranger inspector and Shift-click the named sections. Hit the Copy to New Scratch Pad button, and all of your best clips are thrown together on the Scratch Pad. One last thing is to use the Remove Gaps command under the Events menu so that the clips will butt up against each other and be ready for export.

It's A Wrap

Once everything is done, you've edited your video, composed your music, replaced your dialogue and Scratch Padded the best bits, you'll need to export your finished video. Now, do remember that the audio that's packed into a video file is compressed and not necessarily at the quality you want it to be, so you might want to export a mixdown separately; it all depends on your destination.

The Export Video option is found under the Song menu. You can select the file type and the codec along with the audio compression. The other options are the same as you'd find with audio export, including the ability to play along in real time.

The improved video features in Studio One version 6 are very encouraging, and I wonder if it's something that will develop further over time. If PreSonus added fades, text overlay and the handling of image files, it really could do the majority of what I imagine most of us do in our video software.



The Export Video menu offers a number of options for your final video's file format and quality.

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MIKE LEVINE

Stay on top of your content with Clippings.

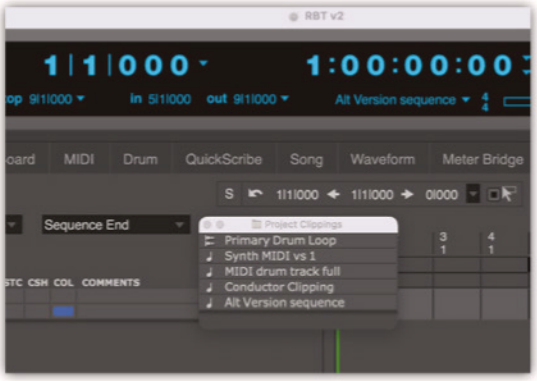
DP's unique Clippings feature offers significant power for organising, saving and recalling Soundbites, MIDI clips, and other track data. With Clippings (not to be confused with Clips, DP's clip-launching feature), you can even save and recall entire chunks (sequences and songs), effects configurations and settings, conductor data and more from inside your DP project.

What's more, any file type on your Mac or Windows desktop can be stored and accessed from a Clipping window inside DP. Clippings are easy to use, versatile and can make your DP workflow significantly more efficient.

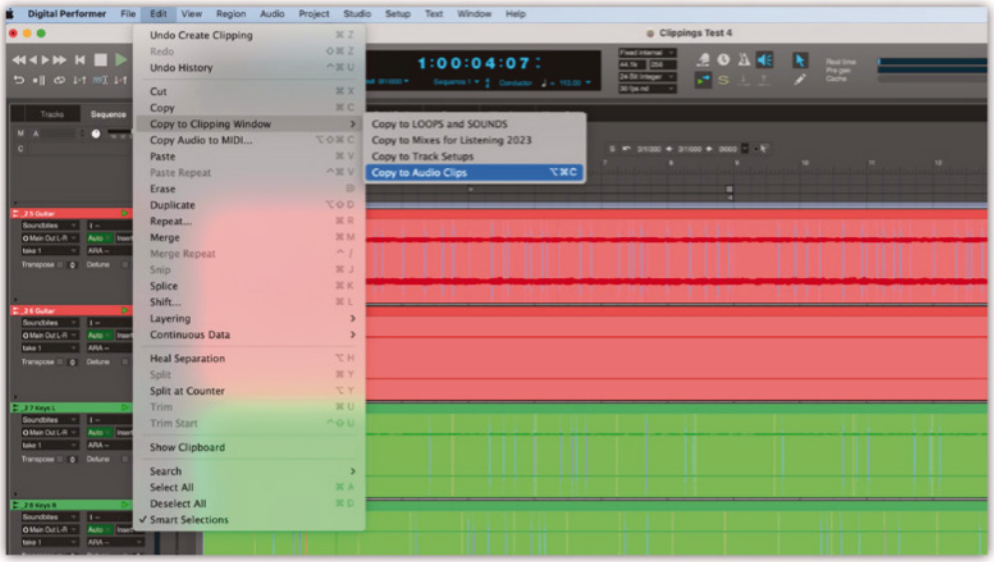
First Things First

Go to Project / Clippings in the main menu to access the Clippings feature. This sub-menu provides options for creating and deleting Clipping windows and changing the icon size for Clippings saved within them.

You can create one of the three types of Clipping windows. A Digital Performer Clipping window is global. The Clippings in it will be available in all of your projects. As its name implies, a Project Clipping window is specific to the project you're working on. Use it for loops, samples, MIDI clips, or whatever you need with your project. A Startup Clipping window is also project specific; in it, you can place third-party applications that you want to open automatically with your project. An example would be lyrics in a Word document for a vocal session, or a file from an ancillary music application you'll use alongside DP in a particular project. For most situations, though, you'll probably opt for Digital Performer or Project Clipping windows.



Clippings windows can hold audio clips, MIDI files and many other types of DP data.



The Clippings feature can be accessed from the Project menu.

The easiest way to create a Clipping is through the Edit menu.

When you save data as a Clipping, you'll see its name and an associated icon for its file type in the Clipping window, but its actual data isn't stored there. Instead, it's copied to a specified folder location, depending on which Clipping window type you've created.

Project and Startup Clippings are stored in dedicated folders inside your Project's folder. Digital Performer Clippings are stored in a specified directory on your computer. On the Mac, it's Library/Preferences/MOTU/

Clippings. On Windows, it's C:\%userprofile%\AppData\Local\MOTU\Clipping.

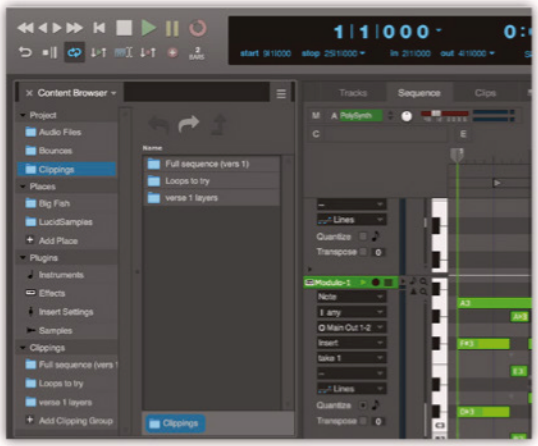
A quick way to get to the folder where your Clipping is saved is to right-click, Command-click or Control-click its title in the window itself, and a pop-up will show you where it resides on your drive.

Saving & Accessing

There are several ways to make a Clipping with track data. The easiest is to select an item from one of DP's edit windows and use the menu command Edit / Copy to Clipping Window, or the key command Cmd+Option+C (Ctrl+Alt+C onWindows). Choose the Clipping window you want to access from the resulting pop-up, and then release the mouse.

Alternatively, you can make Clippings by dragging and dropping files or data into a Clipping window from a DP window. These can include soundbites and sequences from their dedicated windows, soundbites or tracks from the Sequence Editor or Tracks window, conductor track contents, and more.

Clipping windows are pretty small to start with and aren't manually resizable. The only way to increase their size is by loading many files into them, in which case the window will grow vertically to show the name of each Clipping. Once you've created clipping windows, the easiest way to access them is from the Content Browser (in the Consolidated window sidebar). The Content Browser is a great way to create all three types of Clipping window, view all existing clipping windows, and drag and drop clipping data into the DP timeline. Click Add Clipping Group in the Content Browser sidebar to create a new Clipping window. Right-click on Clipping folders in the Content Browser list to rename, open, reveal in Finder, and delete folders. Right-click on the clipping data itself to reveal in Finder. From there, you can rename, delete data, or move it to another Clipping folder.



The most convenient way to access Clipping windows is through DP's Content Browser.

You can also open Clipping windows as separate windows by choosing them from the Project / Clippings sub-menu. Clipping windows won't float on top of other DP windows. As a result, each time you click in a DP window, any open Clippings windows will go to the back, thus disappearing from view. To bring them to the front again, simply reselect the Clipping window from the Project window menu. Active Clipping windows will have a tick mark next to them.

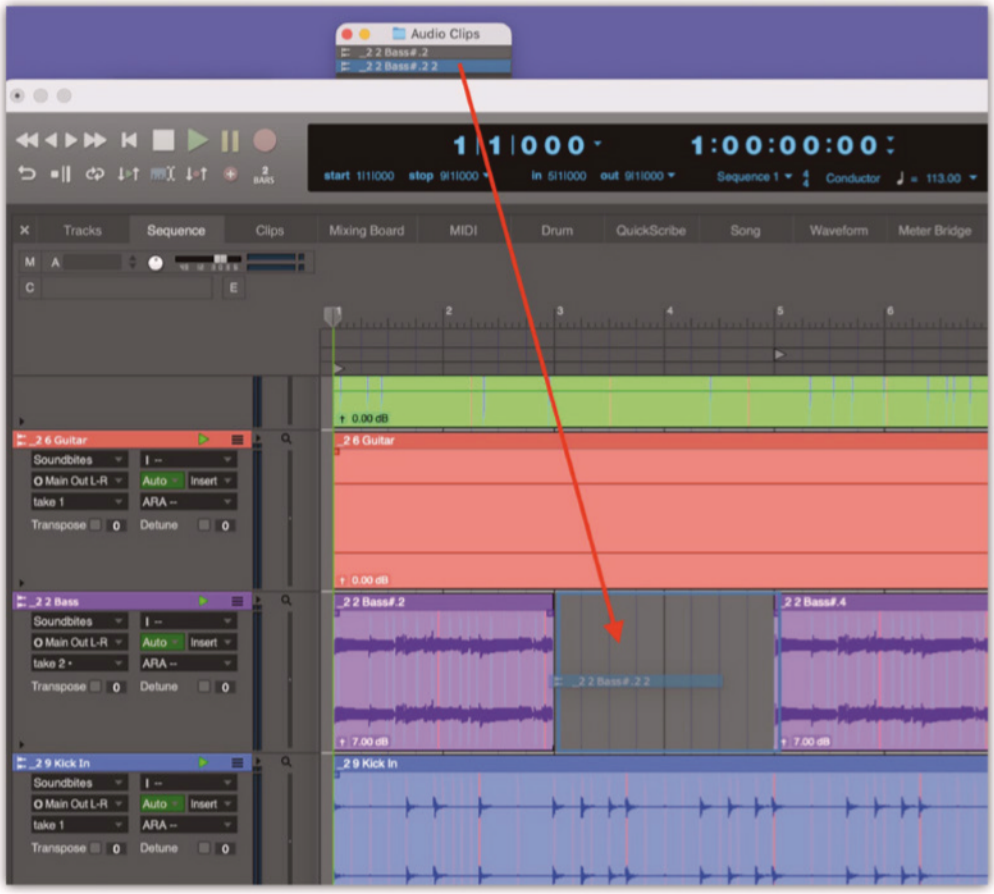
Here's a tip: consider resizing DP's Consolidated window a little bit smaller on your screen and placing the Clipping window to the side of it so it doesn't keep disappearing when you click in the DP interface.

Only Pointing

Another powerful use of Clippings is to open folders on your drive from within DP. For example, if you have a collection of samples, you can access them quickly and easily through a Clippings window. Let's say you have a loop collection that you want available for a song you're working on. Simply drag and drop the folder into a Project Clipping window. It won't copy the data into your project, but when you click the Clipping, the enclosing folder will open.

To place a Clipping (audio, MIDI or other DP track data) into your project, drag it by its icon and drop it in an eligible DP window. MOTU defines eligible as "windows that can accept the type of data included in the Clipping".

If it's an audio or MIDI file and you want to audition it before dragging it in,



Grab the Clipping by its icon and drag it where you want to place it.

» double-click on its name in the Clippings window. If you don't hear anything when auditioning, it's possible that the MIDI instrument isn't correctly configured.

In any Clipping window, you can choose to view the Clippings with small icons, large icons or by name. In a Clippings window with many items, you'll probably want to use the small icon or name setting, to make more items visible without scrolling.

If you choose the small or large icon modes, you must grab the Clipping by its icon to drag and drop it. If you select name, you can drag the clipping from anywhere in its name. However, you can only drag one at a time in Show by Name mode. Use one of the icon modes if you want to drag multiple Clippings at a time. Each Clipping window can have an independent icon mode setting.

To delete a Clipping, select it in the Clipping window and hit the Delete key. To delete a Clipping folder, click on it and go to Project / Clippings / Delete Clipping window. To copy a Clipping from one Clipping folder to the next, just drag and drop it.

Advanced Clippings

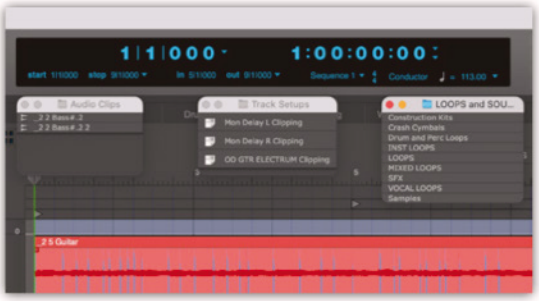
The variety of track data you can save in a Clipping is impressive. For example, you could save a channel or channels from the Mixing Board, complete with inserts, insert settings, sends, panning and volume settings. Unlike the Save Insert Settings command, saving a channel as a Clipping includes the bus routing. Select the channels you want to save in the Tracks window or Sequence Editor and go to Edit / Copy to Clipping Window.

Or let's say you have a favourite set of reverbs, delays and other effects that you like to use on certain songs but don't necessarily want to include in a track template. You could save those tracks — with plug-ins and settings intact — by selecting the track headers in the Tracks window or Sequence editor and copying them to the Clipping window. In a different sequence in the same project, or even in a completely different project, you can recall that set of tracks by dragging the Clipping into the Tracks window or Sequence editor, and they'll appear with plug-ins and bus assignments intact.

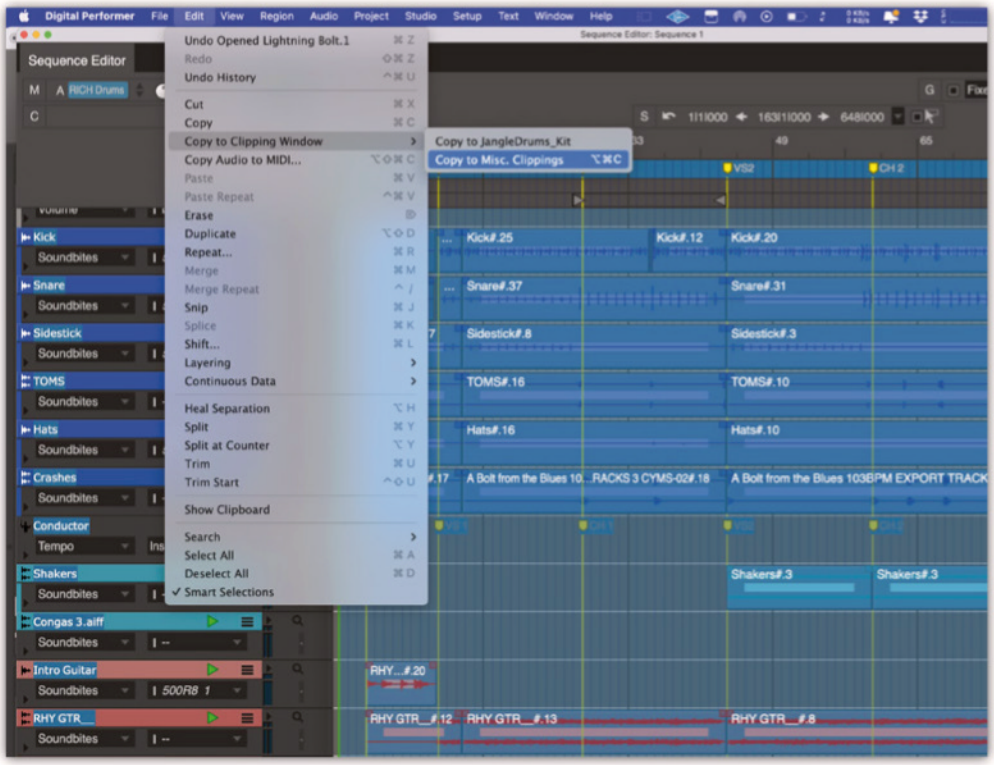
If you enable the 'Create new bus bundles' option in DP's Clipping Preferences, DP will automatically change the bus numbers if they conflict with the existing busses in the target sequence. If you don't tick that preference — or if your Clipping was created before DP 10 — DP won't make the new busses, and you'll have to set them manually.

Combined VIs can be selected and copied to Clipping windows as well. Browsing these Clippings in the Content Browser makes for a very fast way to drag combined virtual instruments into the DP timeline. You can also use Clippings to save entire songs (one sequence per Clipping) by copying the sequence from the Chunks window into a Clipping.

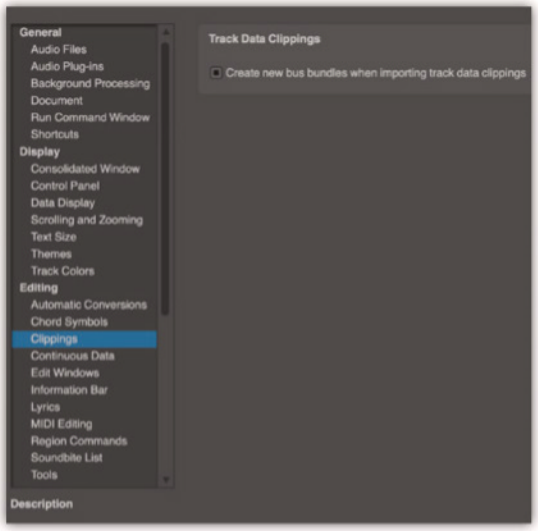
As you can see, the Clippings feature is powerful and deep. It's so versatile that you can configure it to enhance the way you work in DP.



Left to right: Small Icon mode, Large Icon mode and Show by Name mode.



Copying tracks to a Clipping window.



Ticking this preference tells DP to create and renumber busses (where necessary) when a Clipping with routing information is pasted into a sequence.

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With the new Sketch window and app, Pro Tools enters the world of clip launching.

JULIAN RODGERS

At the end of August, Avid gave us a sneak peek at Pro Tools Sketch, a feature slated for inclusion in the (at the time of writing) upcoming Pro Tools 2023.9 software, scheduled for September. Pro Tools Sketch is a non-linear creation tool similar to Ableton Live and Bitwig, or the Live Loops feature in Logic Pro. It adds a whole new window to the Pro Tools UI, which isn't something that happens often. We get additions to the Mix and Edit windows regularly, most recently the tabbed Melodyne ARA view, but the last completely new window I can think of was the MIDI Editor, introduced all the way back in Pro Tools 8.

However, Sketch is more than just a window. It's also an iPad app, which will eventually support the iPhone and Android platforms too. While the free Avid Control app has been available for some time, that only allows control of mix parameters in Pro Tools, not manipulation of audio and MIDI itself. By contrast, the Sketch app allows musical ideas to be worked on in, and transferred between, an iPad and a computer running Pro Tools.

Pro Tools Sketch is featured in all Pro Tools versions, including the free Pro Tools Intro. The working method will be familiar to anyone who has used similar features elsewhere. A grid of cells can be populated with audio or MIDI clips. A library of content is provided, and clips can be previewed in sync with the Sketch from the library before being dragged to one of the cells in the Sketch grid.

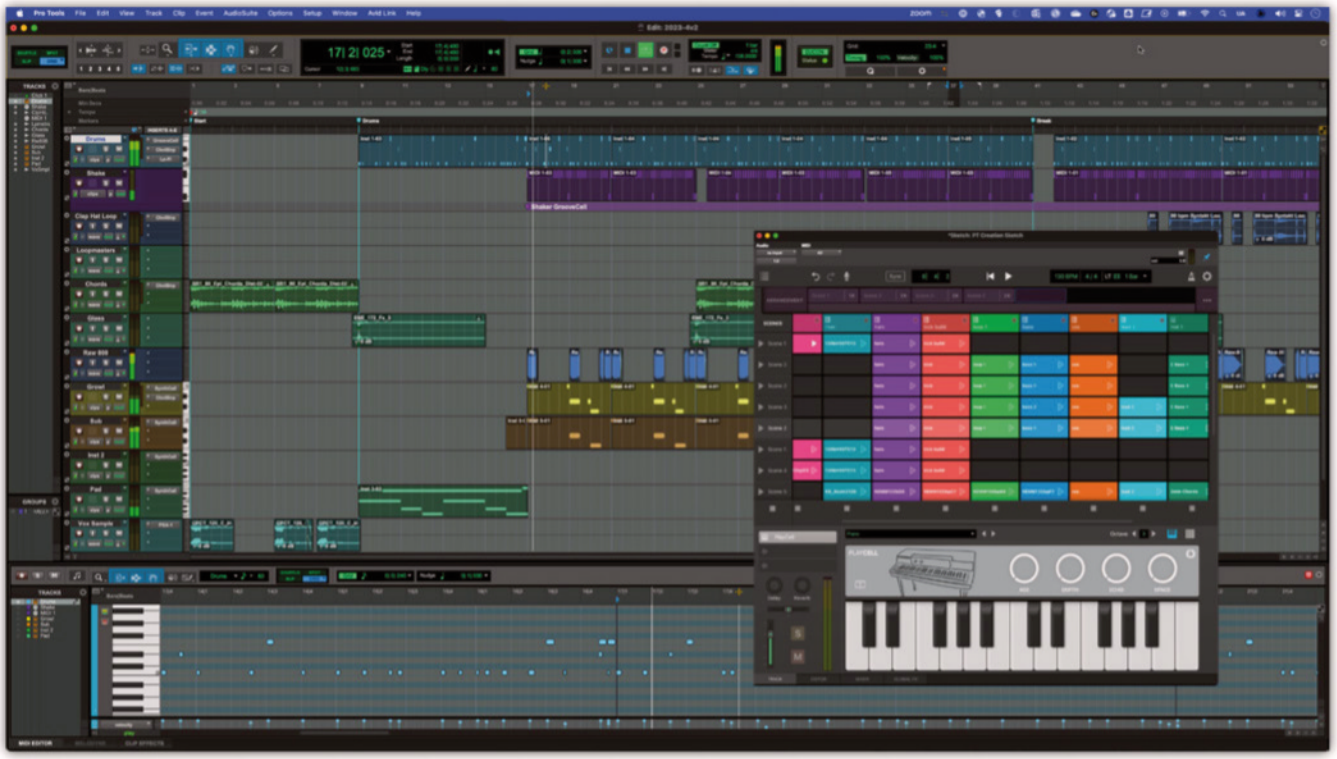
Any individual cell can be triggered from any of up to 16 tracks, and playback loops in sync with the Sketch. Horizontal rows of the grid can be triggered as Scenes, which can be sequenced into an arrangement by dragging them to the Arrangement lane at the top of the window.

Integration

These types of music-creation apps have been available for a long time, and you might wonder whether Sketch offers anything that isn't already possible using third-party apps and tempo-sync tools such as Ableton Link. The difference with Pro Tools Sketch is that you are already working within Pro Tools rather than alongside it. This I see as central to what Avid are trying to achieve with Sketch. The strength of Pro Tools as an editing and mixing platform is well known, but historically, a very common approach has been for users to compose and create in another DAW and move over to Pro Tools to mix the composition. Sketch seeks to bring the composition process in-house and to make it easier for creators to stay within Pro Tools for the whole production. So how does this integration work?

"Having used the Sketch app quite a lot, I can confirm that it is very immediate in use, thanks especially to the ability to mix and match elements freely and to experiment with the order of sections."

Pro Tools Sketches are saved with the file extension .ptsketch. Unlike Pro Tools sessions, which reference media stored elsewhere, Sketch files are self-contained including all content. This makes file exchange simpler. They can be saved to the cloud or transferred via email if you are transferring between a computer and an



The Sketch feature adds a whole new window to the Pro Tools working environment, and is available in all versions, including Intro.

iPad. For archiving, the .ptsketch file can be saved with the session folder. Sketch files are separate from PTX sessions, but they can be 'pinned' to one or more Pro Tools sessions, which will result in the Sketch automatically opening along with the Pro Tools session. This is a nice touch, and the fact that a Sketch can be pinned to multiple sessions avoids unnecessary duplication and makes version management easier.

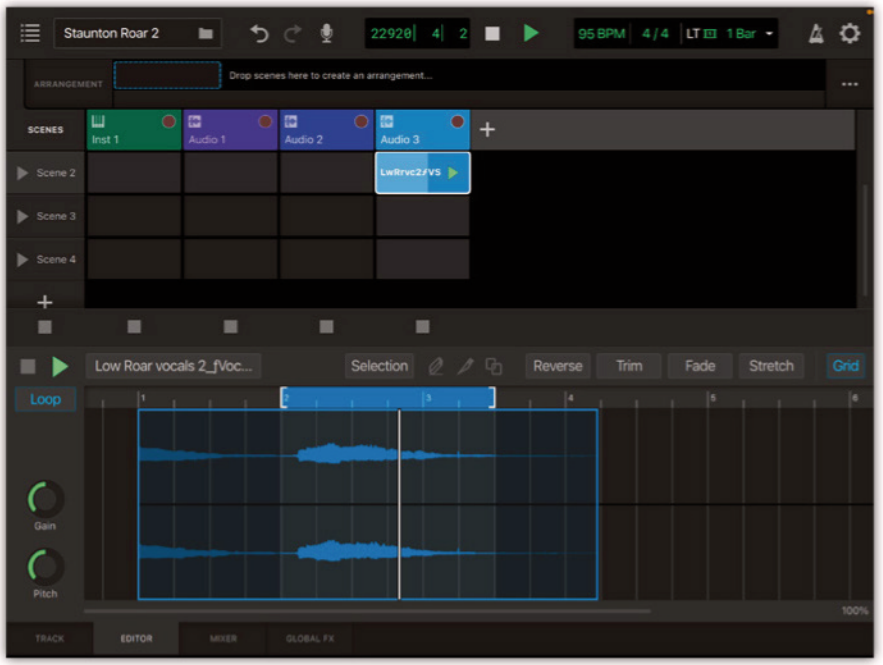
If you are creating on the iPad rather than in the Sketch window in Pro Tools, a completed arrangement can be exported as a WAV file. Sketches can be played back in sync with a Pro Tools session, combining the best of both linear and non-linear workflows, and for more detailed editing Sketch clips, scenes, or entire arrangements can be dragged and dropped into the Pro Tools Edit page, where the full capabilities of Pro Tools can be accessed.

Making Arrangements

The included library that installs with the Sketch app contains content from Avid's Sonic Drop initiative, in which producer Matt Lange creates free monthly 'drops' of content for current subscribers and users with update plans. It's enough to get you started, but creating your own content is simple. Pro Tools is an excellent place to create loops, and being able to work with them on the fly is fast and effective.

MIDI content can be recorded in live, or drawn in and edited. PlayCell and SynthCell are the two MIDI instruments available, and in the mixer tab, any of nine included effects (Chorus, Compressor, Crunch Crusher, Delay, EQ, LoFi, Multimode Filter, Reverb and Saturation) can be used to further shape the sound, alongside reverb and delay which are always available as global effects. Piano-roll editing of MIDI data can be performed, and audio editing, including timing correction and time-stretching, is also available within the Sketch app.

Having used the Sketch app quite a lot, I can confirm that it is very immediate in use, thanks especially to the ability to mix and match elements freely and to experiment with the order of sections — something that isn't convenient in the strictly linear Pro Tools Edit window, where the lack of a convenient place to create alternative arrangements discourages experimentation. Studio One has its Scratch Pad feature, which provides an alternative timeline on which to experiment, but in Pro Tools the best method for alternative arrangements is to create a duplicate playlist for all the tracks in the session and to use Shuffle mode with the All group enabled. It works but can get messy in a session that already contains multiple playlists, and firing off scenes and clips in real time is better while ideas are still being formed.



The Sketch app running on an iPad. Projects can easily be shared between the app and Pro Tools.

Avid have been criticised in the past for catering primarily to professional users, but the introduction of Sketch shows that they are serious about providing tools that are relevant and useful to the next generation of artists, producers and beat-makers too. Sketch certainly enhances the core functionality of Pro Tools to better serve this important group. Pro Tools isn't moving away from the pro market, but they are clearly trying to broaden their appeal.

Pro Tools 2023.9 wasn't yet released at the time of writing, but the tech preview of Sketch, taken in combination with the introduction of Intro and Artist, the release of the Cell series of instruments and the new Sonic Drop programme, show how Avid are working to improve the appeal of Pro Tools to creators. Expect a full review in these pages soon!

STG Acquire Avid

While I'm not party to the internal discussions within Avid, you would have to have been under a rock during August not to have heard the news that Avid has been bought by private equity firm STG. Details from Avid are restricted by their current position as a publicly traded company, which means they have a legal obligation not to make comments that could influence their share price. That of course doesn't dampen the online speculation about what this sale might mean. From the perspective of the customers, the most significant fact which has emerged about the sale is that Avid will return to private ownership and as a result will no longer be restricted by the abovementioned obligations. Using the Pro Tools Sketch technology preview as an example, the reason Avid have shared this preview prior to release is that they are confident that they can deliver it within a known timeframe. If they didn't manage to deliver an important new feature that might influence the perceived value of the company (and by extension their share price), they could be accused of manipulating the market. One of the most immediate benefits to the customers of Avid returning to private ownership is that they will be able to be more open about their future plans, technology, and features which might be coming in the more distant future.

As customers we all like to feel optimistic about the future of the DAW we use, and feeling better informed and more included can only benefit that. The completion of the sale is still in progress and Avid are in a position where they probably can't share as much as they would like to but, without getting too far into business law, having an understanding of why things happen at the speed they do helps contextualise what is happening.

Atmos-capable DAWs Spotlight

Want to mix in Atmos? We take a look at compatible DAWs.

LUKE WOOD

Immersive audio is becoming increasingly popular, and of all of the various formats, Dolby Atmos is undoubtedly the most prevalent. Whilst it has been commonplace in post-production for quite some time, demand for music mixed in the format is on the increase, too, with streaming services such as Apple Music, Tidal and Amazon Music now all offering Atmos content. Dolby state that for a DAW to be suitable for mixing music in Atmos it must natively integrate, or be able to connect to, a Dolby Atmos Renderer, as well as supporting 3D object panning. In this month's Spotlight, we take a look at the options that meet those requirements.

Apple Logic Pro

Version 10.7 of Logic Pro brought with it native Atmos integration, delivering a built-in renderer and extending the mixer's level meters and surround panners to 7.1.4 to accommodate the additional height channels. A 3D Object Panner can also be selected on any non-surround channel, which will automatically configure the source as an object and allow precise spatial panning. A range of the DAW's surround-capable plug-ins were also extended and optimised for Atmos: Space Designer, Limiter, Loudness Meter and a selection of effects processors all now feature 7.1.4 support. Binaural rendering is provided to allow monitoring of Atmos mixes on headphones, with a standard mode joined by a host of options that provide compatibility with the various rendering modes used by Apple Music. There are also speaker virtualisation options for creating mixes that take advantage of the immersive sound features offered by supported Apple devices. Logic can import and export ADM files, although it is not possible to configure downmix settings, and output files can only be delivered in six pre-configured speaker layouts up to 7.1.4. However, it can be used with the Dolby Atmos Renderer in order to overcome these limitations.

$ $199.99
W www.apple.com
W sosm.ag/LogicPro10-7

Avid Pro Tools Ultimate & Studio

Both Studio and Ultimate versions of Avid's Pro Tools now feature support for Atmos mixing, although there is no renderer built in, so they must currently be used alongside the separate Dolby Atmos Renderer application. Both versions can directly route 7.1.2 beds to the software, and also offer a range of channel configurations that exceed the bed limit by using static objects to accommodate speaker layouts with additional height channels — Studio provides options up to 7.1.6, whilst Ultimate extends that support to 9.1.6. When a stereo or mono track is assigned to an object output path or output bus with an immersive channel layout, panning can be carried out using the software's 3D panner, which is equipped with a Theatre mode that displays the location of the sound within a rotatable 3D room. ADM files can be exported using the DAW's Bounce menu, and it is also possible to export re-renders in all of the supported channel configurations. Pro Tools can open Atmos sessions or import ADM files without a connection to the external renderer, allowing edits to be made whilst monitoring a rough mix via the software's Bed/Object Fold Down path.

$ Studio: $299/year or $29.99/month
Ultimate: $599/year or $99/month
W www.avid.com

DaVinci Resolve Studio 18

Although it may seem strange to see a video editing package on this list, Blackmagic Design's Davinci Resolve comes equipped with Fairlight, a powerful built-in audio post-production environment. What's more, if you opt for the flagship Studio version of Resolve, its capabilities are extended to support a range of

immersive formats — including Atmos configurations from 5.1.2 up to 9.1.6. Once a format has been selected in the Fairlight preference menu, tracks and busses will be equipped with a 3D panner, and a 3D Spaceview tool allows users to visualise multiple sound objects' locations. A built-in renderer makes it possible to export Atmos content in layouts up to 7.1.4, and the software can be used alongside Dolby's own renderer application for those who require higher channel counts or more advanced functionality.

$ $295

W www.blackmagicdesign.com

Merging Pyramix Premium 14

Merging have worked alongside Dolby to offer seamless integration between the flagship version of Pyramix and the Dolby Atmos Renderer, allowing users to automatically configure the DAW's I/O routing with a single click. Due to its prevalence in the mastering world, it's no surprise that Pyramix offers a wealth of options for importing Atmos content. It provides full details of ADM files before importing them, and can load single or multiple files into a session with ease; routing to the renderer can be mapped automatically or manually, and there are options for preserving the original configuration or removing unused channels for optimisation. There are also some clever routing options which allow users to work with channel formats that exceed the maximum bed input size of the Dolby Atmos Renderer by using objects to 'emulate' a speaker, all whilst remaining in compliance with Dolby's guidelines. Export capabilities are equally comprehensive, and Merging have included Apple, Universal, Sony and Warner Brothers naming conventions to ensure compatibility with their delivery specs.

$ $2787

W www.merging.com

PreSonus Studio One Professional

The latest DAW to introduce Atmos support is the previously stereo-only PreSonus Studio One, which, as of version 6.5, now includes a built-in renderer and support for channel layouts up to 9.1.6 as standard. As well as introducing surround and 3D panning capabilities, PreSonus have also upgraded the DAW's plug-in set to support the new multi-channel configurations. The latest version of Open Air is kitted out with a new library of 7.1.4 impulse responses, offering a range of built-in spatial reverb options, and there is a new multi-tap Surround Delay plug-in that has been specifically designed with immersive audio in mind. Binaural capabilities are also included for monitoring Atmos projects on standard headphones, and a dedicated headphone channel eliminates the need to change configurations when switching between headphones and speakers.

$ Perpetual: $399.99, Studio One+ Subscription: $19.99/month or $179.99/year

W www.presonus.com

Steinberg Nuendo 12

Steinberg's Nuendo 12 features native integration of the Dolby Atmos Renderer, meaning that users are able to create, monitor and deliver Atmos content in speaker layouts up to 7.1.4 without the need for any additional software. 3D panning is taken care of by the software's VST MultiPanner plug-in, which is equipped with dedicated Bed and Object modes. The renderer also offers a binaural downmix feature for creating an Atmos mix for headphones as well as speaker systems, and there are options for assigning different binaural rendering modes for each channel within a bed. A Trim and Downmix Editor also allows users to specify settings for lower channel count downmixes (including 5.1 and stereo), which are stored as metadata in the ADM files when the project is exported. Although Nuendo supports channel configurations up to 9.1.6, it can only natively render in formats up to 7.1.4, and will need to be used with the external Dolby Atmos Renderer application to export mixes with a higher channel count.

$ $999.99

W www.steinberg.net

Steinberg Cubase Pro 12

As of version 12, Cubase Pro also comes with a built-in version of the Dolby Atmos Renderer, and shares many of the features found in its post-production-focused sibling. It can natively mix, monitor and deliver content in layouts up to 7.1.4, and still offers binaural, surround and stereo downmix options. There are a few restrictions when compared to the implementation in Nuendo, however: Cubase is limited to the use of a single bed, and can only export (not import) ADM files. It also only supports native rendering, meaning that it cannot be used with the external Dolby Atmos Renderer application.

$ $579.99

W www.steinberg.net

W sosm.ag/cubase-12

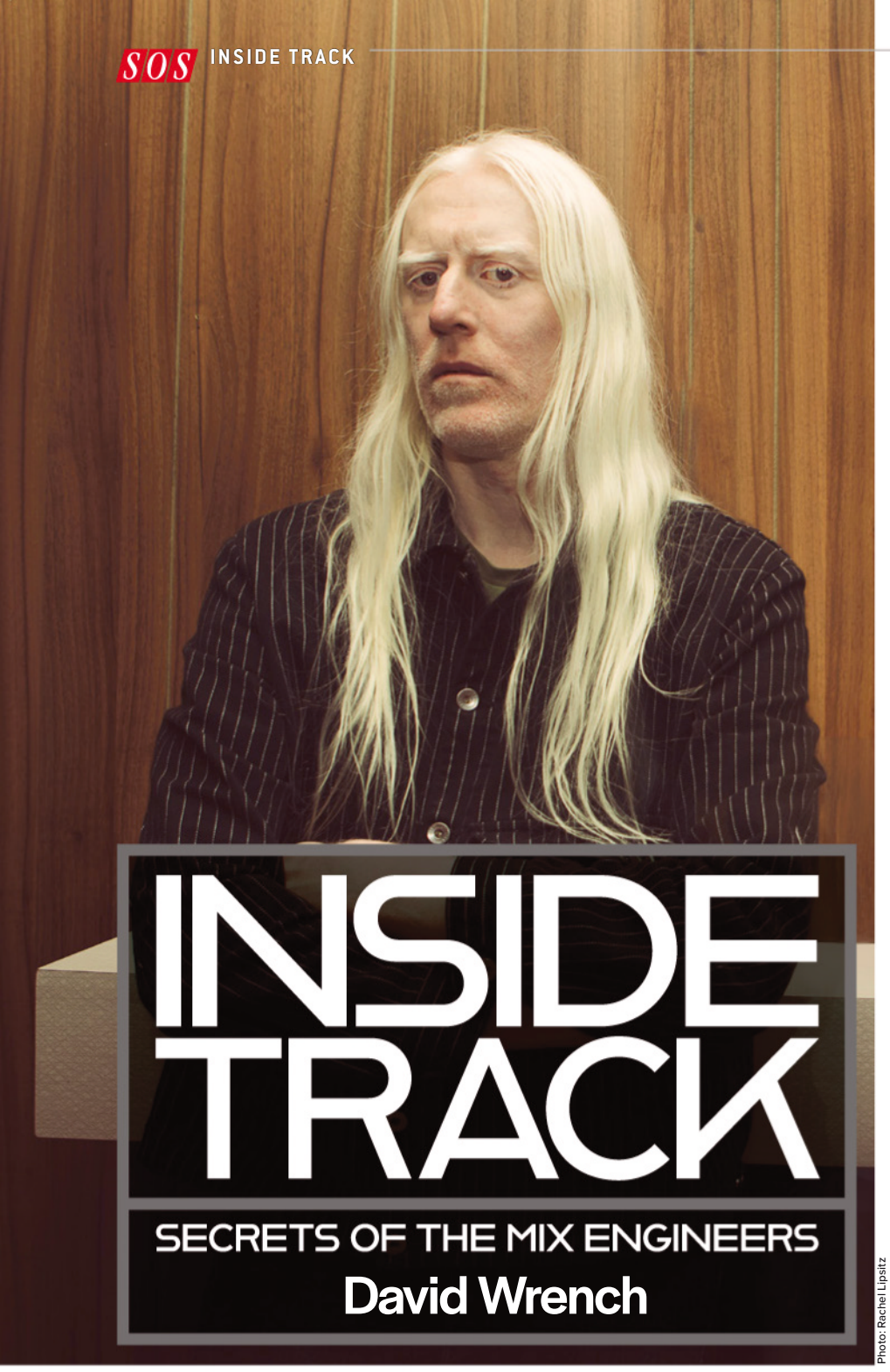
Fieldler Audio Dolby Atmos Composer

If you're keen to give immersive mixing a go but your DAW of choice doesn't currently support it, then all is not lost. Fiedler Audio's Dolby Atmos Composer plug-in not only makes it possible to create and deliver Atmos content using any DAW, but also offers built-in binaural monitoring facilities that mean you don't need to fill your room with speakers first! Dolby Atmos Composer can be placed on the master channel of your session, whilst a second Dolby Atmos Beam plug-in is inserted on individual tracks, enabling their audio to bypass the DAW's mixer functions and be routed to the main plug-in along with 3D panning information. The plug-in can accommodate speaker layouts up to 9.1.6, allows users to export their mixes as ADM files and also offers useful tools such as built-in downmix and trim options and integrated loudness measurement. There is also an Essential version, which provides a reduced set of features at a lower price.

$ Dolby Atmos Composer: $249, Dolby Atmos Composer Essential: $149

W fiedler-audio.com

W sosm.ag/DolbyAtmosComposer



INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

David Wrench

Photo: Rachel Lipsitz

David Wrench's goal as mix engineer was to make Blur sound like themselves!

PAUL TINGEN

As a producer and/or mixer, David Wrench has worked with Ellie Goulding, Glass Animals, Frank Ocean, David Byrne, Manic Street Preachers, Florence + the Machine, the xx and many more. This year alone, he produced and mixed the Pretenders' 12th studio album, *Relentless*, and *Sahel* by Nigerien guitarist and singer Bombino, and, as well as Sampha's second album, *Lahai,* he mixed the whole of Blur's ninth studio album, *The Ballad Of Darren*, which was a UK number one.

The Ballad Of Darren was produced by James Ford, well known for his work with Arctic Monkeys, Gorillaz, Florence + the Machine, and many others. Wrench originally met Ford during a session for a Simian record, at Bryn Derwen Studio in Wales, where Wrench worked at the time.

"This was in the early 2000s, when there were crazy wild parties! It was really nice to have the opportunity to work together again. James is a brilliant producer, and you can see the fantastic job he did on this Blur album. I wasn't there for the recordings, but from listening to the Pro Tools sessions it was obvious that he also had captured the sound of a band playing in a room, and got them to sound like a band. So this is what I tried to enhance in my mixes.

"When you work with a band, first of all you need to work out how it sounds, particularly live. I'm often disappointed when I see bands live these days because they're too slick, it's too polished. Everything is run off backing tracks, everything is triggered, everything is perfect. It's not that exciting. Yes it sounds very impressive but I don't feel engaged in any way. However, when I saw Blur perform, I was blown away. They really sound like a band in the traditional sense, and there are imperfections, which is what makes them so good. It's the same with the Pretenders. They're amazing musicians, and Chrissie is obviously a phenomenal vocalist, just unbelievable. And Bombino and his musicians also are incredible. When you work with people like that, you just want to capture the sound of the band in the room."

Home Brew

Wrench mixes at his own Studio Bruxo in East London. It is a music-making paradise, with tons of synths and other keyboards, guitars, amps, drums, outboard and so on. "I had my studio built to spec, and I have a live room next door as well. There are no big resonant frequencies in the room, as it was purpose-built. I've got a lot of equipment in here, which smears the sound as well. I have two different monitoring systems, one stereo and one Atmos. My stereo system consists of a pair of Neumann KH310 monitors, with the Neumann sub. I know that system back to front.

"I love the Neumanns, I think they're amazing. They're trustworthy, and flat, there are no bumps in the low mids, and they have a tight bass. You just have to be careful with the placement to get the stereo imaging right. They're not fatiguing and they're not flattering, which I really like. You actually have to work quite

hard to make things sound good on them. My Atmos setup consists of 12 speakers, including Unity Audio Rock MkIIs, which were phase-aligned by Dolby.

"I work in Pro Tools with the Avid HD Native Thunderbolt and MTRX Studio, and DADman control software. I run Pro Tools on a MacBook Pro laptop, so I can take it with me when I go travelling. I just got the latest most powerful one, although not all software runs on M2 yet, so I still have to use Rosetta. I have quite a bit of outboard, the most essential units being the Roland Space Echo, and the Eventide H910 Harmonizer. But I'm 95 percent in the box, because you're always going to have to do recalls. It just seems crazy these days to not be in the box."

Embrace Limitations

"I work with a very narrow palette of plug-ins," insists Wrench. "When working in studios back in the '90s, we didn't have 50 bits of outboard gear. You worked with a limited amount of EQs, compressors, delays and reverbs. You'd have a plate reverb, and a couple of digital reverbs. I think having a limited palette of reverbs creates more of a cohesive sound and a space for the track, rather than everything being in separate spaces.

"I'd say 90 percent of my EQ'ing is done on the FabFilter Pro-Q 3, which is just amazing. It's pinpoint precise, but it also has a bit of character if you want it. It has an amazing ability to deal with harsh frequencies in vocals. I also use the FabFilter Pro-C compressor a lot, and the Pro-MB multiband all the time. I'd say that the Pro-Q 3 and Pro-MB are the two plug-ins I use the most to control frequencies.

"I also love UAD plug-ins, they're amazing. The Teletronix LA-2A is fantastic and my main vocal compressor. If I want vocals to have a bit more bite I'll use a UAD 1176, and for EQ that adds character I like the UAD Pultec and the UAD Hitsville, the Motown emulation. The Hitsville has fixed frequency bands, and I quite like EQs that are straightforward like that. I use the UAD Thermionic Culture Vulture on drums or bass to add some harmonics.

"For reverbs, I'd say 90 percent of the time I use the Valhalla

'The Narcissist'
Written by Damon Albarn, Graham Coxon, Alex James & Dave Rowntree
Produced by James Ford

VintageVerb, which is fantastic. It's not expensive and it always does the job. I use the Valhalla Room as well, often on guitars, on a very short setting, just 5 percent mixed in just to give the tiniest bit of space. My favourite delay is the Soundtoys EchoBoy, which is great.

"On the Blur record I used the Soundtoys PrimalTap for pitch delay, with a bit of feedback, again to give things a bit of space, maybe with a slight left to right effect to open it up a bit, and also the MicroShift, which emulates the Eventide H3000. It's a brilliant plug-in. I often put a little bit of that on a backing vocal or a double-tracked vocal, to just widen them, so the whole thing has a little bit of spread to it.

"I'll often use two de-essers, because one often doesn't catch it that well. I find especially if artists have used a valve mic, there tends to be quite a bit of sibilance on records. I can always tell when someone has used a Neumann U47 because the sibilance is quite hard to control. So I use the FabFilter Pro-DS, and then something like [*Oeksound*] Soothe 2 to catch anymore harsh esses. If that's not enough, I'll go in by hand and draw the esses out. I'll be harsher on the backing vocals than on the main vocals so there's not a huge build-up of esses and t's."

Get It Right At Source

In talking about treating vocals, Wrench points out some home truths that are easily forgotten in a world »

Taken prior to the installation of his Atmos system, this photo shows the monitoring position in the Studio Bruxo control room.

» dominated by DAWs in which every performance mistake can be corrected: "When artists have proper mic technique, it makes mixing the vocal so much easier. It's something that's been lost a bit. When Chrissie sings, she moves her head between every word so her breaths are off mic, and she knows to move back off the mic if she's going louder, and she comes in if she wants more power.

"She's learned to do this back in the '70s and it has become second nature. It immediately gives a much cleaner vocal recording. It's the same thing with Damon, he also knows how to do it because he's been recording since the days of tape. Singers like Chrissie and Damon are amazing vocalists who also don't need tuning. When I was working with Chrissie, she'd be like, 'Oh, let me do it again.' There's no resistance to doing another take just to get it right.

"With other singers, when I do tune, I'll use Antares Auto-Tune, but I'll draw it by hand. I don't like hearing Auto-Tune, and when recording I'll push artists, even younger artists, to do another take. I tell them, 'I can just tune it, but doing another take will make it sound better.' Correcting the tuning and timing of performances is everywhere now, but when you're aiming for perfection, you're getting rid of what's special about music. Especially when you're working with brilliant performers.

"Damon is a brilliant performer, and the whole band is brilliant. That's the thing with Blur. They're all characterful musicians. You wouldn't want to break up that timing, because it wouldn't be them, and they'd end up sounding bland. Damon has his own timing, and it's the same with Chrissie. She stretches out the timing and it's what makes her vocals amazing. You get shivers. Can you imagine how a Nina Simone record would sound if it was put in time or in tune? It would sound awful."

The Balance Of Darren

Going into more detail on his mix of *The Ballad Of Darren*, Wrench explains, "I mixed the whole Blur album in 10 days! With bonus tracks, this meant 15 tracks in 10 days, including notes, and back and forth and revisions. The first track I mixed was 'St. Charles Square', which was actually a bit of an outlier, in that it doesn't really sound like the rest of the record. The guitars needed to be quite aggressive, and it has a double drum kit. In the rough mix the two drum kits were panned hard left and right, but this was causing stereo issues, so I muted the bass drum of one of the kits and panned the remaining bass drum central, and then had the top of the kits left and right. It just made the whole thing more focused.

"The second track I mixed was 'The Narcissist', which was more in line with the other material. The trick with that mix was to get the song to build and build over its four minutes. It needed to hit from the start, and then it needed to go up a notch all the time, and build intensity until the end. This involved a lot of volume automation and compression. The compression changes for each part of the song, so I automated it.

"I'm also obsessive about automating volume. I still think 90 percent of mixing is getting volume automation right. I do all this with the mouse. And with the vocals I literally go in and sculpt every word and every syllable. But despite the short time frame I had to do these mixes in, I did not copy my settings from this mix

to other mixes. I like to start every mix from scratch.

"I always reference the rough. I have a Coleman Audio M3PH MkII monitor controller, and I have my mix on one button and the final rough mix that they're happy with on another button, and I'm constantly comparing. I'm always checking that I haven't made it worse. There must be something in there that they like, so I listen to that a few times before I do anything. I listen to what relationships are in the rough mix that you assume the artist enjoys, and then things will jump out to you.

"There'll be a discussion with the producer and the artist beforehand, and sometimes people just want it to sound more open and sparkly, but sometimes people are like 'I've lost track of this, can you help?' With Blur, they'd been working on this a lot and James said 'We need someone with your level of experience to be able to open this up and see what's there.' So in some cases there were quite a few changes. 'Far Away Island', for example, was originally going to be a bonus track, until they heard the mix, and then it went on record, because it had changed quite drastically."

Keep Your Headroom

In the case of the Blur record, Wrench received Pro Tools sessions, which, he says, "had been really well recorded. It makes the mixer's life so much easier. I have an assistant who will, when a track comes in, clean up the vocals and any clicks, look for edits that aren't properly crossfaded, and so on. But in this case it was all fine. I had to do a few bits of mouth de-clicking, but I'm really fussy about that because I compress the vocal quite heavily, so I need them as clean as possible. Because of the amount of compression I use, I also bring the breaths down with volume automation. I like to hear breath in a natural fashion.

"The Blur sessions came with the routing and the plug-ins they had used. The routing was pretty close to what I would use anyway. One thing I checked was to make sure there were no overloads within the busses. Once Pro Tools built in more headroom it started to sound really good, but I do go and check it's not overloading before it's hitting the plug-ins or overloading between the plug-ins. When you follow basic old-school gain structure rules it sounds better. You've got to give yourself more space, more dynamics, more headroom, which allows you to EQ stuff without just adding more harmonics.

"Getting the gain structure right is part of your mix job done. If you're clipping between the plug-ins it just ends up clogging the track and then you do a volume ride on something and all you're doing is distorting it. I always want to be sure that I can do volume rides and not push various busses or auxes into distortion. With regards to the plug-ins in the Blur sessions, I had to get a few licences to make sure all the plug-ins were working, but they were generally using a very similar palette of plug-ins to what I use."

Take It To The Limiter

Wrench uses his mix session of the first single from *The Ballad Of Darren*, 'The Narcissist', to illustrate his mix approach. "My mix process is that I'll first go through things individually. I check the drums, add »

Seeing Is Believing

"The part of my mix process that's probably the oddest," says David Wrench, "is that once I think I'm there, I put the mix on loop and I'll lean back or lie down and just space out, listening to my Neumanns at low volume, for 20 minutes or so. The reason is that I have strong synaesthesia, and I can actually see sound as shapes and visuals. When I'm really concentrating on a mix and have visual stimulation, like when looking at the screen, it's not as present. But when I lay back and listen with my eyes closed, the synaesthesia is very strong.

"I can really see things, and I can tell literally from how the mix looks to me as to whether stuff is working or not. If it looks messy, it is wrong. If my vision is clogged with anything, I know that's what I have to sort out in the mix. I can see that two things are in the same space in the image, and they're fighting, and I need to open them out a little bit. It's a great technique and quite handy. I keep tweaking until I get a really clear visual of the mix in my mind's eye. It's almost like how people describe meditation. You're completely focused on what you're doing, you've got a clear mind. It's the state you aim to be in, as a musician, where you're locked in, and time stops to mean anything. It's amazing."

» the bass in, then the guitars, synths, and finally the vocals, getting a general good balance before I start doing all the volume automation. Once all that's in, I get my mix bus to a good place. The Pro-MB and the SSL G-bus [*on the master bus*] are a very important part of the sound. I always have the SSL at a ratio of 2:1, fastest release there is, often quite slow attack. I just want it to tame things. It may be doing 4 to 6 dB of compression at most.

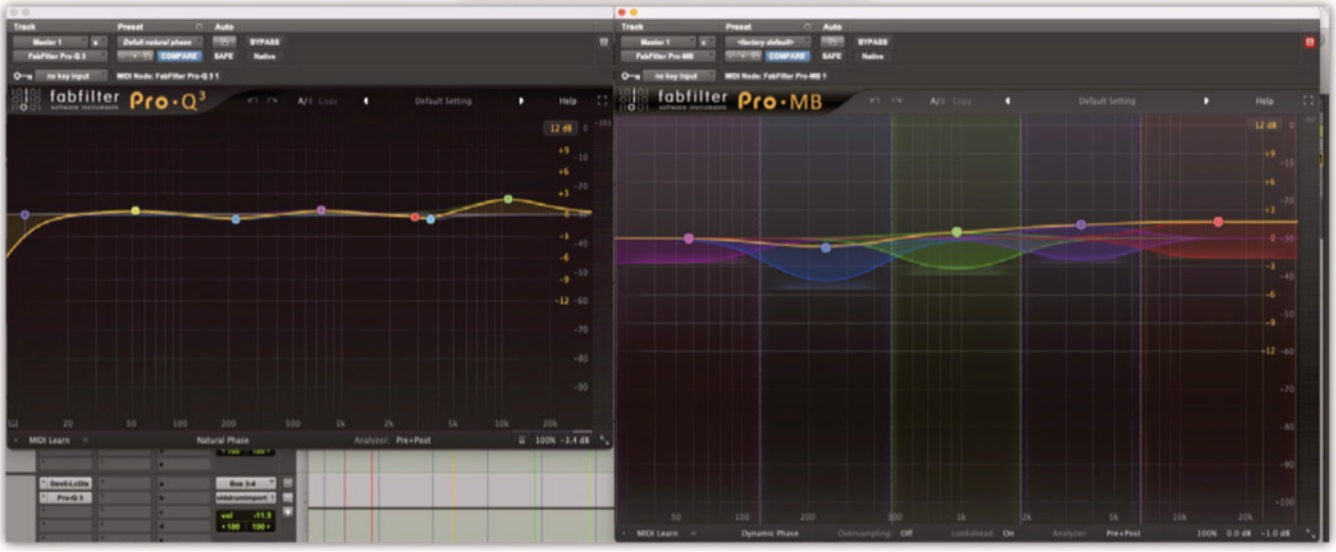
"At that point I'll also switch the [*FabFilter*] limiter on and I'll work with the limiter on for the rest of the mix. When doing volume automation I need the limiter in, so I know what level I'm at, always. I want to know that the mix can survive the mastering process and still sound right. It means working a bit harder to make sure the transient attacks and other things can pass through the limiter without problems.

"Once I have the master bus engaged, my main focus is sculpting the vocal. I'll do a lot of automation so the vocal is really clear. I'll go through line by line, word by word. That's the most time-consuming bit. It can take a good two or three hours just getting that right. During this time I'll listen to the track with everything in. Often in a really brilliant mix, if you solo something it may sound a bit odd, but that's fine because you're not listening to it in solo, you're listening to it in the context of a mix.

"You listen to what is special about each part in relation to the others. What's this guitar part doing? How do I get the essence of this guitar part to cut through without getting in the way of whatever else is going on? Sometimes you have two things happening in the same frequency range, so you need to separate them out, or intertwine them, depending. You can separate them out using different spaces, or panning, or you do notched EQs so they are not quite in the same frequency space. Those are the main things you can do if something is clashing.

"Obviously the vocal is the key to everything. Everyone hears when a vocal sounds wrong. Vocals have to sound good, so I'm obsessive with EQ on vocals. I have an aesthetic of how I want the vocal to sound. I want it to be warm, clear, characterful, and in the right space. I want to be able to play it really quietly on a small radio and hear every word, but not for it to sound separate from the track. I also address occasional frequencies in the voice that need taming, either with a bit of multiband or with notches from an EQ. Once I've got the voice sounding great, I know where I've got to EQ the guitars around or the piano, or whatever. Nothing can get in the way of the voice.

"Damon's a brilliant singer, like Chrissie's a brilliant singer, and they project well. But I still go in and do the details at the end of words, maybe lift it up a touch. Actually I thought Damon's voice sounded amazing on this record.



Once a basic balance has been set up, David Wrench then turns his attention to the master bus, setting up his EQ, multiband and stereo compressors so that he can mix 'into' them.

The Studio Bruxo live room is full of quirky and interesting instruments.

He's got this beautiful warmth which has actually got better with age. When you're working with people with iconic voices like that it makes the job a lot easier, because you've just got to make sure that the character comes through, and they sound at their best."

The Rhythm Section

"I would say that I probably put the drums louder than they've ever been on a Blur record, and the bass is quite clear. But Alex James' bass is picked and isn't subby. I wouldn't try to force that into being something that it isn't, by adding sub or in other ways artificially strengthening the bass. What you hear is what was there, and I made the bass sound as good as possible within the context of the band.

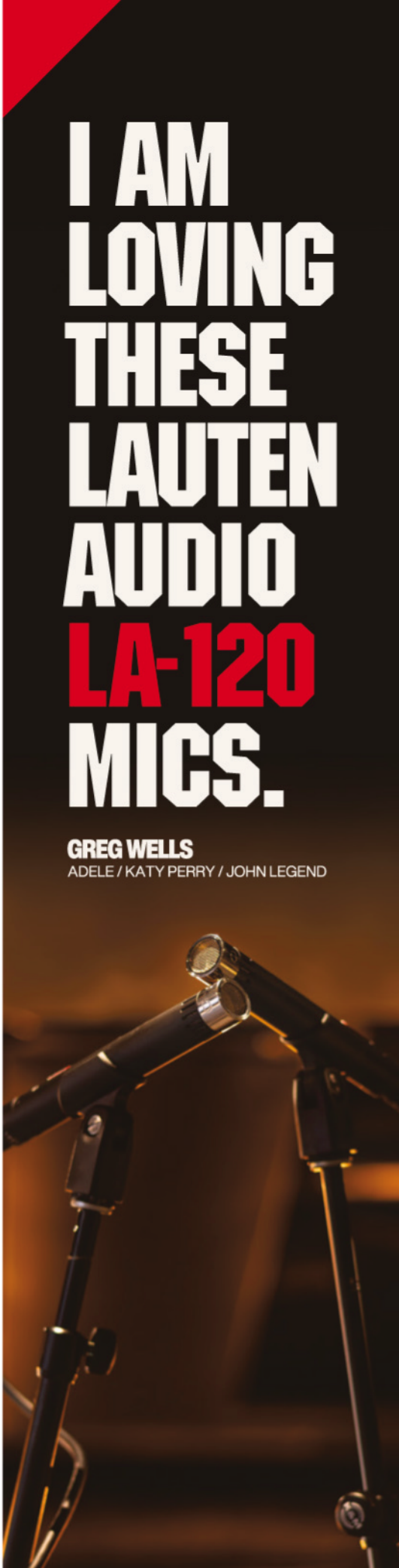
"What's interesting with Blur is that the bass is actually very melodic, almost like a countermelody to the guitar line a lot of the time. The bass doesn't sit in the frequencies of a heavy bass as you have in much modern music. Instead it's above that, it's in the low mids and actually some of it is in the treble. And during the mix I was actually adding more treble to the bass, so you're getting more of that counterpoint. That allows you to sit the kick drum underneath, so that's weighting the record down.

"With more contemporary hip-hop type music and R&B, the bass is underpinning everything and often it's not even bass but an extended and tuned 808, and the traditional bass drum part is much more clicky and above it. When you've got both in the same frequency range it gets messy, and you've got to decide who's going to get the low end, who's going to get the mid, and who's going to get the top end.

"People often will be like 'We need more bass!' when the bass is already sort of maxed out, and what you actually want to hear are the higher harmonics of the bass to make you feel the bass a bit more, and that's what I did on this album. In general I'm very wary of adding sub to melodic bass parts, because you often get very weird harmonics and discordant stuff. But I will sometimes add subharmonics to a kick drum if I feel it's not reaching quite low enough to anchor the song down."

Personal Perspectives

Once Wrench's mix is complete, he takes off the master bus limiter before sending his mix for mastering, which in the case of both the Blur and Pretenders albums was done by Matt Colton. Looking at the enthusiastic reception of these albums to two bands that have been around for a while, Wrench reflects: "People change as they get older, and from that perspective, I think the lyrics on the Blur album are brilliant. They're very personal and from the perspective of someone in their 50s who has been through a lot. The same comes across with Chrissie. And these voices are relevant. Record companies have been guilty of amplifying just certain voices. But now there are many more diverse voices, which is the best thing ever. I want to hear people's stories, and I want to hear something I have not heard before. I want to hear what people from all over the world, from different ages, and all sorts of backgrounds, are thinking and doing. That is what is exciting. Because it is a way of learning about human experience from all sorts of perspectives."



Roger Manning

Keyboard Player & Songwriter

From co-founding cult band Jellyfish to writing with Brian Wilson and touring with Beck, Roger Manning has enjoyed an extraordinary career — and a remarkable collection of vintage instruments.

TONY BACON

Ocean Way Studio, Los Angeles, 1992. The band Jellyfish are recording their second album, *Spilt Milk*. Jellyfish core members Roger Joseph Manning Jr and Andy Sturmer are assisted by several guests, one of whom is Jon Brion. While he's there mostly to play guitar, he's also brought along and set up a couple of keyboards. Not just any keyboards: there's a double-manual Mellotron and, even more intriguingly, an Optigan.

The Optigan was a '70s keyboard produced by toy manufacturer Mattel (and yes, they were also to blame for Barbie dolls). It played samples from 12-inch optical discs in styles described as Nashville Country, Polynesian Village, Bossa Nova Style, and so on. The resulting sounds were wonky enough to be interesting in an alluringly lo-fi kind of way. And Roger Manning was hooked. "The bug just bit me! I mean, I had never physically performed on an Optigan. I knew what they were, I knew what sounds they made, but I'd never sat behind one. I was like 'Oh my God. I'm a poor musician like everybody else, but I've got to figure out how to get my hands on some of these things.'"

Cut to 2023, and Roger is proudly showing us round his music room where he is now surrounded by, at the last count, 132 of these things. Not 132 Optigans — that would be foolish — but 132 different hardware keyboards and modules. He points out a Roland Jupiter-8 and a Sequential Prophet-10 here, a Siel DK80 and an Elka EK22 over there, a Kurzweil 1000SX and an Oberheim Matrix 6R, a Casio FZ-1 and a Yamaha CS80. And many, many more.

"Obviously, I'm a keyboard enthusiast," he says, laughing. "And of course I've got all the soft samplers and everything in the computer, so if a song really wants a trumpet sound I can go get one. But I love being able to come into this room and work within the keyboard sound palette, everything from early synthesis through digital synthesis through sampling. I love it all!"

Beck & Call

We'll dig deeper into Roger's treasure trove later, but first we should examine the Beck connection. He's worked with Beck since the late '90s, most recently on a six-week tour that took up all of August and a chunk of September 2023, as well as a sizeable rehearsal period beforehand. It was, he reports, a real treat working in this band alongside musicians he hasn't played with for eight years: Jason Falkner (guitar, BVs), Justin Meldal-Johnsen (bass, BVs), Joey Waronker (drums) and Ian Longwell (percussion, Ableton), with Roger on keys and BVs.

The practicalities of modern touring meant that his hardware collection stayed at home, sadly, and Roger's setup for the Beck tour was necessarily sleek and flexible. His primary tool was a Nord Stage 2 EX 88, not only useful for its acoustic and electric piano sounds, but also supplying the luxury of 88 weighted keys to trigger sample libraries. Manning mentions their live version of Beck's 2002 song 'Paper Tiger' as an example. "Essentially, it's a rhythm-section song with very elaborate string arrangements, and I'm performing all the strings off the sample libraries, but via the Nord. That really helps me as a keyboardist, because I like having the weighted action for expression."

For many of the Beck performances Roger was also deploying sound effects: handclaps, odd needle drops, and so on. "And very often we wouldn't be playing those rhythmically — I'd have them over on my left hand, on the bottom of the keyboard. In the middle I'd have keyboard stuff: organs, pianos, harpsichords, Clavs, and so on. And while there's a lot of guys in the band who sing, Beck might want reinforcement, so then I could trigger background vocal phrases, different harmonies, things like that."

Free Space

Another consideration in performance is that Beck likes to be free to vary the tempo. If the energy demands they speed up, Beck wants to be able to go with that. "And as we know," Roger says, "10 years ago you had to commit. You didn't have that option."

The key to this facility today is the band's Ableton operator, Ian Longwell, founder of Spacebar Society. "We've known them forever," Roger says, "and we were »

On Beck's recent tour, Roger Manning triggered a huge variety of samples from his Nord keyboard.

» one of the first bands they started working with. They run a bunch of computers with Ableton on them, which not only allows for backing tracks, in the rare events that we want that, and more specifically allows for very elaborate samplers, but also this tempo thing. Let's say I'm performing a background vocal part or a rhythmic loop that plays along with the band, a light percussion thing that has a little bit of harmonic information, maybe a guitar rhythm, some brass, or something. I'm playing that, and it's a crucial part to one of Beck's grooves."

If the band speeds up as Roger's playing that loop, naturally the loop needs to speed up as well. "We've embedded the loops with click," he explains, "which is being fed to the drummer, or anyone else who wants it. Ian, our Ableton operator who thankfully is also a drummer, sits there and taps out the bpm with one of his hands while the drummer is meeting Beck's demands to speed up or slow down."

As soon as Beck found out he had the option to vary tempo in real time, it happened a lot. Roger grew up playing drums, too, and he still takes pride in his sense of rhythm. He says that to keep on top of this fluidity in tempo, he has to really work with the drummer to make it feel right. They don't want it ever to sound at all robotic, or jagged, or jerky. He and Joey have played together quite a lot over the years and have become adept at relaxing into grooves. "And then Ian is the third part of that triumvirate," Roger adds. "It takes great practice, but now we're all pretty good at it."

As well as Beck, Spacebar work with many other artists and record companies. "They called me in last year because 5 Seconds Of Summer were going to do a very special live and recorded performance at Royal Albert Hall," he recalls. "Ian and his team were in charge of hiring a string section and a choir and making sure they all had arrangements, and then adding to what was normally a four-piece band with auxiliary people — there was a percussionist, plus I came in to play additional keys and sing background vocals, and one of the singer's girlfriends did some duets — an insanely elaborate situation. I was happy to be a part of it, but it was really fun to be a fly on the wall and watch everything that Ian and his team did besides just provide the sonics of this Ableton-generated world."

It's a world where Roger is happy to play his part, even if it's not his top personal choice. "No," he says, laughing, "not my preference, personally — because it's not my training. I prefer the organic thing. My training was a lot of improvisational music, lots of jazz and so forth, so I like being with other musicians who say: OK, here's a melody and the chord changes. What are we gonna do with it?"

Electrification

Roger grew up in California, and soon he was banging pots and pans and boxes, »

Cooking The Books

When Jellyfish broke up, Roger Manning began working with the band's guitarist, Eric Dover, for the project that produced Imperial Drag. Roger's quest to own an Optigan was about to peak, a happy development that led to him making two albums with Brian Kehew as The Moog Cookbook. If you fancy the idea of tongue-in-cheek multiple-classic-synth covers of 'Whole Lotta Love' or 'Even Flow' or 'Born To Be Wild' and more, these are the records for you.

He met Brian through the Recycler, a Southern California newspaper dedicated to secondhand stuff. "He was selling an Optigan, I went over to his apartment, he showed me how it worked — and I noticed he had an ARP String Ensemble, a Polymoog, a couple other pieces. I said, So you're a keyboard player? He goes, Well... not really. Said he was mostly a guitarist and an engineer, but that he loved those kinds of instruments. Anyway, we became friends, and over the next year and a half we recorded these drum-machine all-synthesizer arrangements of popular Top 40 songs of the day — for our own amusement, but also to enjoy using a lot of these relics that we both loved to death."

The line-up of magnificent keys on the Cookbook albums (*The Moog Cookbook*, 1996, and *Ye Olde Space Bande*, 1997) included Farfisa organs, that Optigan and the String Ensemble, Clavinets, many Moogs, naturally, a Roland System 100 modular, and several more. Brian was teaching a class at Cal State Dominguez Hills and got to use its studio after hours — and that studio had not one but two ARP 2600s. "That was nice," Roger says, laughing at his understatement, "and I'd found a Jupiter-8 and my Memorymoog by that point. So we had no shortage of great toys to play with."

I wondered if they laughed as much while they made the Cookbook records as I do now when I listen to them. "We did! Brian was busy as a producer and engineer, working with lots of generic rock bands, I was with my rock band, and Moog Cookbook became our mutual escape. We shared a lot of the same sense of humour and infused that kind of lightheartedness into the arrangements. The records were very much inspired by those Switched On type records of the '60s and '70s, whether it was *Switched On Bach* itself or that whole trend. When the Moog modulars came out, unless you were Emerson, Lake & Palmer, a lot of people didn't know what the hell to do with them. So they would just redo classical works: 'Let's see if I can get my Moog modular to imitate a bassoon.' That's great — but we found that kind of thing boring. We liked when a lot of those arrangers started doing Switched On hits."

He points to arrangers such as Hugo Montenegro, whose *Moog Power* album of 1969 was in the vanguard. "The stuff they were doing was super-inventive, very animated. Obviously we were listening to Rick Wakeman blast a Minimoog solo, which was very enjoyable, and Patrick Moraz or something, but we loved what some of these arrangers were doing with the synthesizer — so inventive, so creative. And often a lot of those sounds had a comedic overtone to them, and we'd never really heard that before. We really tried to capture that spirit in the Moog Cookbook arrangements."

Apart from being a lot of fun to make, a benefit of the Cookbook records came when they were noticed by the French duo Air. "They reached out to us to do a couple of remixes, and then eventually Brian and I became part of their backup touring band for their first album, *Moon Safari* [*1998*]. It was hard enough not to fall in love with Air if you were a keyboard player — we just thought so much of them — but they were on the other side of the world, so we thought there was no way we were ever going to meet. And it was around that time that I was getting introduced to the Beck people, because they'd just lost their keyboard player, and it was actually the Moog Cookbook that intrigued Beck the most. I don't even know to this day if he's heard any Jellyfish."

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A small selection of the 132 keyboards currently housed in Roger Manning's studio!

» inspired by an uncle who played drums in a surf band, Davie Allan & the Arrows. His mother had other ideas and insisted on piano lessons, which at first he had no interest in. Mum won, of course, especially as his grandparents had donated a piano to the family, and Roger soon found himself sitting in a local house waiting for his lessons to begin.

"My teacher had put out a stack of *Contemporary Keyboard* magazines in the living room, and I would see these long-haired freaks in glitter capes — Rick Wakeman playing Moogs and shit. Somewhere in my mind I knew, well: this makes some kind of futuristic sound that the trumpets and the violins and bass guitars can't make. Otherwise this guy wouldn't have four or five of them all around him. And of course I couldn't tell a Mellotron from a Hammond B3 from a Minimoog from an RMI. That would come later."

By the time he went to high school in 1980, he was lapping up as much prog, synth pop and fusion as he could find. "My friends and I were trying to learn the Herbie Hancock and Chick Corea and Tony Banks parts by day, and then come night-time we were going to punk rock shows in San Francisco — lots of art, lots of tear-down-the-barriers. There was all that, not to mention the incredible jazz that was getting electrified and going in all directions. And those were also the very early days of sampling — bands like Depeche Mode were using Emulator IIs, Kate Bush was using a Fairlight. It was the most exciting thing in the whole world."

A few years later, Roger had his first synth, a secondhand Roland JX-8P, and an Ensoniq Mirage sampler. "Guitar music ruled the day in the latter half of the '80s, which I liked as well, but it never occurred to me to take up the guitar. Oh, I learned my barre chords, but I was like: you can do everything on keyboards! Any sound you want, grab a keyboard. I was the sort of auxiliary guitar player in the band, making tough sounds, distorted Clavs, aggressive synthesizer sounds, crazy sample sound design, all those things."

Beyond Jamming

Then the songwriting bug hit. When he thought about the music he admired the most — like Thomas Dolby's solo records and his production work for Prefab Sprout — he found that what drew him in was the quality and ambition of the songwriting, in addition to the way the records were made. "I realised that I could jam quite well on people's music, but I couldn't write an original hook or a strong verse to save my life. I would have been about 19 or 20, and I literally stopped practising. I needed to figure out how to write a song! And that's all I did, trying to write an idea not only that I liked, but that I wasn't embarrassed to play for my friends.

"That was a long, painful process," he adds with a smile.

You just have to learn the craft, he says, and start transcribing your heroes. "Bands like XTC were at the top of the heap for me, the Elvis Costellos of this world, a lot of '60s British invasion stuff, on and on, but the point is that keyboards became my writing tool. If the song ended up as a more guitar-oriented pop song, that was fine with me. I just started using keyboards as a way to navigate. And anything I heard in my head, I could get on the keyboards."

This was an advantage over his friends who wanted to play guitar. "A lot of them would learn their basic folk chords and be off and running. Well, that's fine, except you're going to be at the mercy of how many chords you know. Your songs are only going to be as inventive and creative as your basic chord vocabulary. Having studied keyboard since the age of five, the sky was the limit. My knowledge of harmony was quite advanced, so it wasn't going to be I-IV-V for me. I have no problem with three-chord rock — some of my favourite songs are three-chord rock — but there's a benefit in having that power in your back pocket. I was very grateful for that, because I knew that a lot of my friends didn't have it."

Fishing Expedition

Roger co-founded Jellyfish in the late '80s, and he recalls that as he and Andy Sturmer began work on their first album (*Bellybutton*, 1990) they both wanted to bring in the sounds of yesteryear. "Hammonds in new ways, Wurlitzers in new ways, Mellotrons in new ways. Synthesizers were a big no-no at that time — that was the height of grunge — and we were shy of those elements. But by the second Jellyfish record we started bringing synths into it, and samples, too."

Spilt Milk (1993) is a beautiful piece of pop splendour, a detailed creation carefully designed from the ground up that provoked much praise from contemporary musicians

and producers. It was recorded in 1992 around Los Angeles, and Roger remembers how they had grown since *Bellybutton* not only instrumentally but also as arrangers. "We could hire a real string quartet to come in and play parts that we would arrange on samplers or keyboards, or if we preferred the scratchy Mellotron string sound, we would use that, or we would combine the two, or... it was whatever we wanted. That can be a double-edged sword, of course. But Los Angeles was a great place to be, loaded with some of the best players in the world. So we would bring guys in to play, say, harmonica, guys who'd been playing on people's records for 40 years. Just a very exciting time."

One of his favourites on *Spilt Milk* is 'He's My Best Friend', which he describes as an uplifting slice of '60s/'70s-style singer-songwriter pop, very Harry Nilsson in its approach and production. "We had fun arranging that, because it's lighthearted but also has a beautiful, almost Simon & Garfunkel lyricism. I spent a lot of time arranging those clarinet ensembles — completely out of my realm, but you

throw yourself into these crash courses in arranging. How could I assemble all my musical education and come up with something that's fun and inventive, but doesn't get in the way of a lead vocal? It's the constant arranging conundrum."

The cinematic 'Russian Hill', meanwhile, features the unusual combination of Bruce Kaphan's pedal steel guitar and Roger's string arrangement. "That was my very first real string arrangement," he says with pride. "I spent a long time on it, and it was a 20-piece orchestra, so it had this larger kind of grand Hollywood sound to it."

The studio team was the same as it had been for *Bellybutton*: Albhy Galuten and Jack Joseph Puig. "Jack was coming more from the sonic side of things," Roger explains, "and Albhy more from the traditional Tom Dowd, Phil Ramone, Quincy Jones-style producer, that understanding of everything from the song to 'Your demo idea works here, but this is why I think it may not work and serve you best here.' We trusted those guys like big brothers, mentors, parental figures. We were very lucky to have been introduced to them. We knew that no matter how great or unconfident we were about certain songs and our demo'ing, they had our backs, that whatever song we tackled, this was the perfect four-person team to do the job."

There are clear influences resonating around the vocal arrangements on *Spilt Milk*, but Roger has an interesting perspective on the connections. "It's an area that fascinated the hell out of me, and Andy too. A lot of my vocal arranger heroes, whether it's Brian Wilson or Richard »

» Carpenter or Fleetwood Mac or Queen, a lot of it comes from keyboard players. Freddie's a keyboard player. Brian's a keyboard player. Richard's a keyboard player. And I think the vocal work appealed to that part of my arranging sense, too. It was a lot of trial and error — Andy and I would have to work it out, eventually, and we lost and gained band members along the way, but thankfully we found very talented guys who could sing. We started to do it on *Bellybutton*, but in my opinion our dreams and goals were perfected on *Spilt Milk*, which to me is much more the realised Jellyfish sound. I feel we really did what we set out to do by that second album."

Was Not Wis

Jellyfish had some big fans who recognised good work when they heard it. One such was Don Was, and the result was that Roger got to meet one of his all-time musical heroes. Don had worked with Roger and Andy on a song they wrote and recorded for Ringo Starr's 1992 album *Time Takes Time*, and that in turn led to an encounter with Brian Wilson.

Don was working with Brian and figured Roger and Andy might be a good fit for a songwriting collaboration, so he arranged a meeting at a Beach Boys warehouse in Santa Monica. Roger recalls walking into a workroom with a piano. The atmosphere was unsettling. As well as Brian Wilson, there were four or five people running around who appeared to be his assistants, and it was unclear who to address. Roger hadn't known if Brian would come along with ideas, so he and Andy showed up with a rough idea for a song.

"I sat down at the piano and played for Brian, with everybody else just standing around. I probably have never been more nervous in my life," Roger remembers. "Especially because we had no lyrics. I had to mumble melodies. I believed in the ideas, I felt they were strong, but it was still very awkward."

He ploughed on, playing and mumbling through a verse and chorus a couple of times. "And then, of course, you usually go into some kind of breakdown or bridge, or a solo. So right where I stopped, I looked up at everybody and said: 'And that's where...', and Brian leapt to his feet and goes: 'That's where we put the bridge!' He sat down next to me at the piano bench and started playing chords and mumbling ideas. I'm like, oh my God, somebody pinch me! Is this really happening? And right there, on the spot, he came up with an idea. I barely remember what it was, but the song was in progress. It certainly wasn't like it was finished, but it was enough that we could all tell that Brian was inspired, excited, and seemed to be cool with us. So we were all looking at each other going well, can't wait to get together and do this next time."

Roger and Andy went back to finishing *Spilt Milk* and stayed in touch with Don. "And it was during those next three or four weeks that Don and Brian parted ways. We have no idea why. But that was the end of that. We were like: 'Noooooo...!' The Lord giveth and the Lord taketh away." Roger later resurrected the core ideas to write 'Wish It Would Rain' on his 2006 solo album *Solid State Warrior* (also known as *The Land Of Pure Imagination*).

Whistle Stop Tour

Before we leave Roger in his room full of synths, there's time for a look at a few more of the prize items in his collection. "There's a part of me that I guess is a bit of an antique collector," he admits as we poke around. Oh look! Is this what I think it is? "Yes, I recently acquired a PPG Wave. They're hard to find, and even when you do find one it can be pricey. I was turned on to a gentleman here in Los Angeles who actually refurbishes them. It was not cheap, but it's mint condition, and I know that it's going to work for a very long time. But also, one of the joys over the years for me has been to not spend a lot of money on these things. I like treasure hunting. I like finding ones that are broken, and I know several great techs here in Los Angeles who'll resurrect them for me. I do still like trying to find deals."

There's something about a physical object in today's world of virtual this and

An ARP 2600, Korg Delta and Yamaha CS80 in Manning's studio.

Another view of Roger's studio, with his favourite Roland Jupiter-8 and Sequential Prophet-10 in the foreground.

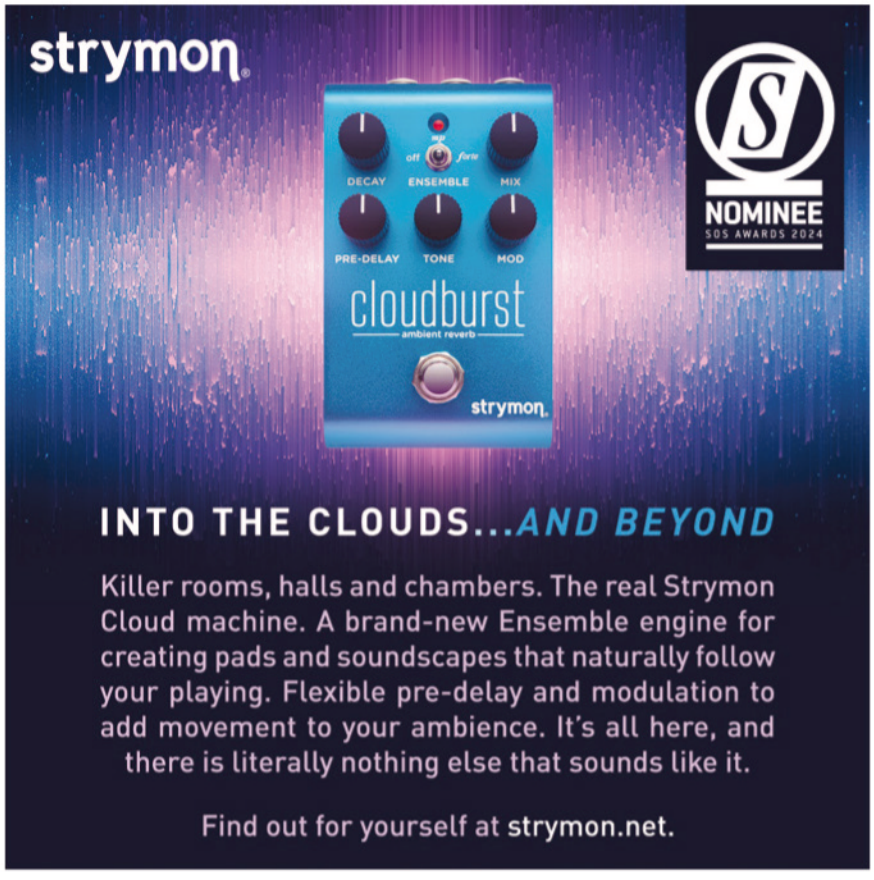
that, isn't there? "A hundred percent. I love the tactile, tangible aspect of grabbing the knobs. Also, I've purchased modules in the past four or five years that I never would have touched in the '80s or '90s, because they were fairly average, you know? You had to menu-dive just to start extracting the possibilities. Now, of course, there's all kinds of tech nerds around the world who make interfaces and controllers with knobs to get inside of these things."

Appropriately, we stop by a single-rack Yamaha TX81Z. "It's just a four-operator FM synth," Roger says, "and my friends and I used to laugh at it back in the day. Like, who the hell would buy this piece of shit? But the Stereoping guys in Germany make a great multi-controller for that thing. And the best part is that it's just like vintage synthesis. One of the joys of the early synths was that you didn't have to be a musician. You turned the thing on, you started moving knobs, not knowing what you're doing, and you'd suddenly got a crazy sound. That's how you have New Order and Cabaret Voltaire and Heaven 17, all these wonderful bands. Depeche Mode, too — I mean, Vince Clarke is a master, but back in the day, they were learning too. You start turning knobs, and suddenly you're getting these crazy, screeching sounds out of this Yamaha. This is fantastic!"

OK, now we seem to be at the Oberheim section. "Well, I'm always looking for Oberheim SEMs, and I have a two-voice and a single, but the Oberheim sound is probably my favourite synth engine of all time," Roger says. "I can't even describe it, but it just vibrates my cells the right way. It's super organic and natural-sounding to me. So, I found a broken OB-X, which I'd been longing for forever. I put a good chunk of money into it to get it rehabilitated, but the beautiful thing about those is that they're so fairly straightforward that my tech was able to source the parts, and he said, 'This is going to take me a while, but I know exactly what to do.'

"So, yes, within the last year I've acquired an OB-X," he says, "and I couldn't be happier about it. But, you know, I'm sat here in this room, and I don't really need anything." Well, there's need and want. Two different things.

"Exactly..."



Ludwig Göransson

Film composer and producer Ludwig Göransson has earned a stellar reputation for sonic world-building.

SONAL D'SILVA

Ludwig Göransson has two guiding principles when starting work on a new composition: he never reaches for the same instrument twice to put down his initial ideas, and he likes to limit the sounds he uses. They might seem mutually exclusive — blue-sky experimentation on the one hand, and a limited palette of sounds on the other — but when he explains further, it becomes clear why it works. "If you always put down your ideas the same way when you're inspired — say you immediately reach for the guitar — there is muscle memory at work and your fingers are just going to go in a certain way. Same with piano. So I try to approach every project like I'm doing something for the first time."

Starting Point

A quick examination of his film credits corroborates this. For his Oscar-winning score for the film *Black Panther*, the Swedish composer literally went to a completely different place: the continent of Africa, where he collaborated with local musicians and learned the intricacies of their musical customs. "Music in Africa is different from music in the rest of the world; it's not a performance, it's part of the culture and the language. All the different drums mean something — you only play this type of drum when there's a wedding, you only play that type of rhythm when it's a funeral, you want to play this rhythm on the talking drum when someone is coming back from having been gone for a long time... so with *Black Panther*, it was always super important that we stayed true to the culture and used those instruments in a way that they would actually do in Africa." For the sequel, *Black Panther: Wakanda Forever*, he went to Mexico City, immersed himself in the local music scene, and spent time with a music archaeologist to uncover the sounds of Mayan music that have been gone for over 500 years.

Photo: Magda Wosinska

Ludwig Göransson's recent work includes the scores for *The Mandalorian*, the *Black Panther* films, *Creed*, and the Christopher Nolan-directed epic, *Oppenheimer*.

For *The Mandalorian*, he went online, bought a set of five recorders, and freestyled in a room by himself for hours until a soundscape emerged. "I chose the bass recorder because I had never played it before and it looked really cool. I started playing it and it was just beautiful; I ran it through some delays and some reverbs, and it sounded a little spacey. So for every project, I am trying to do something different with instruments and sounds."

For his latest, the Christopher Nolan-directed biopic *Oppenheimer*, the starting point came from one of the only suggestions the director gave him: that the violin be at the heart of the score. Göransson took that note and ran with it. "Chris Nolan wanted the violin to personify Oppenheimer's character. I think he was interested in it because the violin, being a fretless instrument, is very responsive to performance; you can change the tone of it within a split second and go from something beautiful and romantic to something neurotic and horrific within a really short amount of time. Something that we also talked about was that violins and strings have been used so much in film, often as a horror effect with string clusters, for example, so [*we thought*] 'How can you take those types of effects but play it in a beautiful tone?' We also played around a lot with the dynamics of a traditional orchestra, going from a solo violin to an octet, and then to a string ensemble. You can grow throughout the story and sometimes make it very intimate and personal, and sometimes very operatic."

In the early stages of experimentation, he explored how to use the familiar instrument in an unfamiliar way with his wife, Serena McKinney, who is an accomplished violinist. "We started experimenting with the performance of the violin and microtonal glissandos, and one of the first things I wrote was the *Oppenheimer* theme, which you hear throughout the whole film."

Make It A Double

Oppenheimer marks the second collaboration between Göransson and Nolan (the first being 2020's *Tenet*) and to hear Göransson tell it, it was a truly collaborative and deeply involved effort. Even before a frame of film had been shot, based on their early conversations, Göransson had written three hours of music for the director, so when Nolan began editing, there was no need to use a temp score. Once editing was underway, the two met each Friday to watch the movie from start to finish and listen to the progress on the score. While instruments like the violin, piano, strings and harp form the bulk of the early part of the score, as the story moves forward, the music evolves to include electronic elements.

"Most of the time, the theme is played by a solo violin, but when you see Oppenheimer having a breakthrough, having these vivid images or dreams, there's a synthesizer that takes over the melody; it almost sounds like a distorted horn ensemble, and we also doubled it with a brass ensemble to make it feel both real and synthetic — I call it 'worldising'. Apart from that, I also brought in a lot of synthesizers and modern production. To me, those sounds foreshadow the impending doom."

He describes a cue, titled 'Ground Zero', that underscores a pivotal moment in the movie. "All of a sudden it's not theory, it's not scribbles on a paper any more; it's an actual bomb, they're putting it together, the whole thing is up in the air, and the »

Ludwig Göransson's studio is a mix of real and virtual. His computer setup is based around Cubase, Live and VSL software, while the adjacent live room houses a variety of acoustic and electronic instruments.

» music takes a big shift there. There are only three elements in that cue: a throbbing, pumping bass that makes you feel like it's your heart beating; there are radioactive nuclear sounds, a crackling in the speakers; and then there's this little metallic ticking sound. It goes from lush, romantic music to having this really intense sound-design feel." Throughout the film, Göransson used sounds that he recorded especially for the project and incorporated them into the score, he explains, "like nuclear sounds or apparatus from that time, [*that I made*] sound modern by manipulating them in Ableton Live".

Elsewhere, string glissandos were doubled with parts played on the Minimoog, the draw for Göransson being that you could play it live, with the pitch wheel. He elaborates, "If it's a string quartet playing a glissando with four different voices, you can play it live with the Moog synthesizer too; I like how the performances are slightly different and how you can make it feel human but using the synthesizers as an orchestra." The doubling technique can be heard all over the score, to achieve different effects, he says, "At times, we have parts played by a string quartet and overdubbed with four different synth voices; you can double it and have them weave in and out of each other so you don't really know if it is the real instrument or if it is something else. People are unable to put their finger on what the sound is coming from, which is also something that's very interesting to me — you can create something that maybe sounds familiar, but there's something different about it, and it sounds new."

The Freedom Of Limitations

Another limitation that the duo imposed on the score was that it would feature no drums at all. The thought process was clear: "We didn't want to connect Oppenheimer with a military feel, and the feeling of no drums [*in the score*] was really good! Also, there were some really important sound-design elements in the film, like foot stomps and bombs, for example, and having no drums and no big hits in the score really gave those elements more room."

While the sound palette was minimalistic, the ideas were free to be as over-the-top as they liked. The cue 'Can You Hear The Music?', for example, has 21 tempo changes (starting at around 80bpm and ending at 360bpm), and was played in real time by live musicians. Göransson reflects on the evolution of the piece: "When I first sequenced it in Cubase, the piece didn't have a tempo change; it went between 16th notes and triplets. However, when we started recording live strings playing it, you could tell, when we did the metric modulation and it switched to triplets, that the performance of the strings started to change and you really started feeling like it was in threes instead of in fours. So then we came up with the idea to have everything be 16th notes and to have double as many tempo changes instead."

When it came time to record the piece with an orchestra, Goransson had his doubts about pulling it off. "I thought we were going to have to record it four bars at a time because I didn't think it would be possible to have humans perform this live. But the musicians we were recording are the best in the world; they sit in the studio »

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Göransson and director Christopher Nolan on the set of *Oppenheimer*.

Photo: Melinda Sue Gordon / Universal Pictures

» for eight hours a day with a click in their headphones. So we came up with a way to give them the new tempo two measures before the tempo changed on the page, and they were able to play with that and that's how we recorded the two-minute piece in one live take."

Hybrid Arrangement

Göransson's process of recording the complex arrangement in real time, instead of programming it on a computer, mirrors Nolan's well-documented affinity for painstakingly handcrafted in-camera special effects. Göransson, however, isn't interested in being an analogue purist, and his studio hosts a variety of digital tools to assist his hybrid workflow. "When I work in film, I sequence in Cubase and ReWire it with Ableton, and I have all my MIDI in Cubase linked with two computers that I use for Vienna Ensemble Pro, where I host my orchestra samples. Then I have a third computer where I only have the picture, the sound effects, the dialogue, and the temp track and this computer is also where I print all of my audio."

Since shaping and manipulating sound is a big part of his process, he lists a wide range of plug-ins that are frequently used in his productions, ranging from ubiquitous third-party ones like FabFilter Pro-Q 3 and Pro-C 2, which he uses for EQ'ing and compressing, to humble but effective stock tools like Ableton's built-in Auto Filter and EQ. He explains, "They're pretty great when you just want to program; it's super easy to draw lines. I like it especially because when I'm running Ableton as ReWire, obviously I can't open up a plug-in that doesn't belong to the program."

Also in his arsenal are two plug-ins from the Los Angeles-based software company Output: the granular effects processing plug-in Portal, and Movement, a multi-effects engine which, as the name suggests, adds rhythm and movement to any sound. For reverb, modulation and chorus effects, he enjoys the Valhalla plug-ins, and for delay, he says, "it's always EchoBoy by SoundToys." He's a fan of SoundToys in general and is all praise for Crystallizer (an Eventide H3000-inspired pitch-shifter and reverse echo processor), and Decapitator, for distortion. Göransson's use of the SoundToys plug-ins can be heard extensively on the *Black Panther* score, with Decapitator being used on drums and on the flute in Killmonger's theme, and EchoBoy on vocals. XLN Audio's RC-20 Retro Color, Antares' Auto-Tune ("on vocals, obviously, but you can put it on other instruments too") and iZotope's Trash and Ozone round up the list.

While the plug-ins are used to shape tone, Göransson brings up the physicality of composing several times, and his studio is set up with a variety of instruments to allow for that. The composer is a talented multi-instrumentalist and any limitations that exist are self-imposed in an effort to foster creativity. "The last place I want to go sit down [*when I'm inspired*] is at the MIDI keyboard. I also try to not be in front of a screen because you can put in a lot of time and a lot of your ideas into the computer but it's never going to give you anything back. At least with an instrument the body resonates — it's talking, it's singing if there's something happening there." A wall of guitars and a Eurorack setup underscore his point.

A great example of his process can be heard in the theme for *The Mandalorian*. It starts off with the bass recorder, which in turn inspires a tom drum rhythm that leads to a piano part, and then an acoustic guitar line; a Mellotron part fills out the arrangement, and the piece finishes with delicate motifs played on a Fender Rhodes. All of this still leaves room for an orchestra to give it that classic Star Wars feel. Another great example of his method can be heard in T'Challa's Theme from *Black Panther*, which is built around the talking drum, a traditional instrument from West Africa, whose sound can be manipulated to mimic human speech. A chorus of talking drum players was underscored with the 808 playing the same rhythm as the talking drums to emphasise the beat and create a powerful low end for the piece.

The Low End Whisperer

Göransson excels at shaping the all-important low end, the bugbear of every »

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» new composer and producer who has spent many a long night trying to turn a muddy, messy low end into a powerful and propulsive foundational element. Göransson has figured it out and he's happy to share his insight. Unfortunately, he says, there's no magic button. "You are never going to solve any issues in the mix after you've written your song. What's important is to always think about the bass sounds that you are going to use when you're adding them to your composition — they have to match what is already in [*the arrangement*]. You can't write it down on a page, hope it's going to be good, and that the mixer is going to [*solve*] it. You need to have production in mind when you sit down and write parts. When you're about to add bass to your composition, listen to it over and over again, and try to imagine, in your head, what the bass sound is that you want before you put it in; that's a good way of feeling and knowing what's right.

"Also, with experience comes the knowledge that it's not about having 10 great sounds [*in your piece*] — if you have three great elements, you can see how far you can take them. That's going to be much more important than just messing around with as many sounds as you have. It's difficult on your computer and on the keyboard because you have access to everything — a hundred grand pianos, a hundred synthesizers, and 20 different drum kits; you have an unlimited sound arsenal, but I like limiting myself with sounds and limiting myself with ideas."

The other key to a good bass sound, he says, is to have "good rapport, good communication and trust with the person who's mixing your music. Chris Fogel has been mixing my film scores and we talk a lot about how I want it to sound. The demos already need to sound good and then what Chris Fogel does is see how he can elevate the demos. When I work on the low end in my studio, I'm sure it's blasting out loud but then what Chris can do is probably take out certain frequencies that are not going to sound good in a theatre."

With A Little Help From My Friends

With that he dispels the myth that a composer is a lone genius, crafting sounds that move millions all by himself. In fact, collaboration comes up repeatedly in the conversation as something Göransson desires and is good at. "What I think is most important is that you have to get out in the world and get out of your comfort zone. When you're young, and especially if you're studying at a music school or a conservatory, it is important to find other voices and start working with other artists, composers, or musicians that have a different music education, play a different instrument, or work in a different genre of music from you. A lot of people just tend to stay in their building [*at university*], but you can just walk over to the other house, go in there, knock on the door and someone's going to open it. I think that collaboration across different genres and different backgrounds is very important. I started out in rock and metal and then got into jazz, classical music, hip-hop, and all these different kinds of music that I love."

He brings it up again when asked about how he navigates the world of big-budget studio films where every aspect of the film is decided by committee, a process that some people argue stifles creativity and individuality. Göransson, who started out working in TV and on indie films, was thrust into the biggest of the big leagues — a superhero movie — with 2018's *Black Panther*. He says he didn't find that particularly jarring and there's a good reason why: it all comes down to who your collaborator is. "I'm very fortunate to have some really great collaborators, whether it's Ryan Coogler, Christopher Nolan, Jon Favreau, or Childish Gambino. The people that you work with the most, it is important that you feel like you can trust them, and that they can trust you, because when there is a big budget, when there are a lot of people involved, there are going to be a lot of voices and a lot of people talking. That can be stressful if you're working with someone you don't really know, or maybe with someone who doesn't really understand music, and they have their producers and then executive producers in the studio giving them notes all the time. You can be in a tricky position, but for me, I've been very fortunate to work with collaborators that have been very respectful to me and know how they want their music."

He views his other role — that of Grammy-winning music producer and co-writer who has worked with artists like Haim, Adele and Childish Gambino — as deeply collaborative as well and, therefore, not particularly different from his job as a film composer. "It's not that different, because whether it's a rapper or a singer or a director, you're still trying to understand the vision of the artist. The biggest difference, when I produce music for another artist, is that you work a lot more closely with someone in terms of spending your time with them. I like both aspects of it: I like sitting alone in my studio when

Ludwig Göransson's work often straddles the border between composition and sound design. His score for *Oppenheimer*, for example, includes the sound of a Geiger counter.

I'm writing music for a film, but I also like producing an artist's album or song where we can sit together in the studio and talk about music; that's the really important part of it, the social aspect of making a song together."

His association with Childish Gambino, the musical alter-ego of actor/filmmaker Donald Glover, goes back to when he landed his first TV show gig, providing music for the show *Community* on which Glover starred. The two hit it off and went on to collaborate on several Childish Gambino projects, including the 2016 Grammy-winning hit *Redbone*, and 2018's visceral *This Is America*, which won all four of the Grammys it was nominated for, including the highly-prized Record Of The Year and Song Of The Year awards.

He reflects on the success of the song, "What a moment! I first worked with Childish Gambino when we were 24 years old and we released [*the mixtape*] *Culdesac* online from a small little apartment in downtown LA. I remember feeling like, 'OK, everyone's going to hear this!' But when you work on a project [*like* This Is America] that becomes such a huge thing almost overnight... I remember the day the video came out and the song came out; it was special."

He is similarly quietly enthusiastic about working with Adele, one of the biggest names in pop music, leaving one to think that he measures his success by some personal, artistic metric. The pair worked on two songs for her last album and the highlight, according to him, was "just getting into a room with Adele and understanding why she is so brilliant and who she is; it was the first time I had worked with her. Socially, she is incredible to hang out with, but also writing music together with her and hearing her come up with these ideas and lyrics... She has an input in every aspect of the song, she really knows what she wants; she has such a good gut feeling and such trust in her musical self. So both of us were in the studio, we were focused on getting things done, and the musical conversations that we had were just really incredible."

New Perspective

It isn't just diplomacy and people management in the film and music business that is helped by having good collaborators. In Göransson's case, he has embraced how it influences his approach to the very process of scoring a film — watching the visuals set to his music by a director changes his perception of the music itself. "For example," he says, "when Chris Nolan starts editing a film, he might take a piece of music that I wrote and put it in a scene where I would not have thought to put it; or I wouldn't have thought to write that if I had that scene in front of me. I think that's really interesting because a lot of times if you just start to write music to the first cut... you're sitting there with a keyboard and you say, 'OK, I need to write a cue for this scene.' If the scene is one and a half minutes long, you might end up writing a one-and-a-half-minute piece of music — you let the picture control what you're doing instead of the opposite.

"Sometimes it's good to just turn off the cut, think about what you just saw and how it felt, and then write; you don't base the music on the timing of the cut (which is probably going to change 10 times anyway), and you don't think, 'OK, well, I'm going to dip down here for the dialogue and then go up there...' I work like that too sometimes, but I think when you're trying to get at the emotional core of the movie — the big themes that you know you should get — that for me happens, most of the time, when I'm not watching the picture."

Start From Scratch

In 15 years, Ludwig Göransson has gone from being a music school graduate to winning three of the four major entertainment awards — he has two Emmys, three Grammys and an Oscar. His body of work is eclectic, and if there is a pattern running through the diverse soundscapes he builds, it is that he likes to create his own sound libraries for every project. "I don't want to use sound libraries or loops that I've already heard a thousand times; what's fun for me is to create new sounds from scratch. It's important to talk to the director about the world that he or she is trying to create, and to see how we can create that world together using ideas from the script or from those early conversations. They are creating this world from scratch, so why wouldn't you do the same with the sounds? You do it with melodies, chords, and harmony, but also, especially with technology and computers today, you can really push things, through sound."

This has been a practice he has put in place since his first feature film, the award-winning indie drama *Fruitvale Station*, where he recorded and used the sounds of the BART train that is central to the story, and it has continued in every project since. He recalls, "On *The Mandalorian*, we recorded musicians that played something organic and then took those sounds and loops that we made and ran them through modular synthesizers and effects. You can take familiar sounds, familiar instruments, and completely turn them on their head. For *Creed*, I knew that I wanted to record a full day of a boxer training in the gym, so [*sound designer*] Christopher Lane, who I work with often, brought his sound team out there and recorded the boxer. I didn't know what I was going to use the sounds for, or how I was going to put them in the movie, but we ended up taking those sounds, manipulating them, and making them into beats."

On *Tenet*, where he worked with Christopher Lane and Anthony Baldino, he used field recordings of sirens and truck sounds that were pitch-shifted and side-chained to provide the low, rumbling bass for the cue 'Trucks In Place'. The score mirrors the movie's themes of reversing time and inversion, he explains: "The way I was manipulating the sounds on that movie was definitely a step further in terms of inverting the sound — for example, I recorded a live drummer playing a cymbal crash, reversed it, had him listen to that reversed recording, then asked him to emulate the reversed cymbal crash, and then I reversed that again. There are so many ways you can experiment with creating new, unique sounds. That's what I do; I'm trying to come up with my own world."

The M People

How beyerdynamic's M Series Microphones Are Made

We visit the German factory where beyerdynamic's classic microphones are still handmade in exactly the way they always have been.

SAM INGLIS

With centenary celebrations planned for next year, beyerdynamic have a good claim to be the longest-established manufacturers of audio equipment in the world. Founded in Berlin by Eugen Beyer, their initial products included cinema loudspeakers, and as early as the 1930s they were already making microphones and headphones. After the Second World War, beyerdynamic relocated to Heilbronn, north of Stuttgart, where they remain to this day.

Eugen Beyer was a good friend of Georg Neumann, and the two had an informal agreement whereby their respective companies would focus on different design approaches. Neumann devoted himself to perfecting the capacitor microphone, whilst beyerdynamic specialised in electrodynamic devices: moving-coil microphones, headphones, loudspeakers and ribbon microphones. The company's track record for innovation in this field is second to none, and they still employ two full-time staff devoted to prototyping and exploring new designs.

The path from ground-breaking design to established classic was also a relatively short one for many beyerdynamic products. The iconic white, rectangular DT 100 headphones, for example, have been studio staples for decades, and few other microphone manufacturers can list as many genuine classics in their current ranges.

A rebranding exercise in the 1990s saw some of beyerdynamic's studio microphones added to the TG ("tour group" or "touring gear") line, with the M 88 and M 201 for example becoming the M 88 TG and M 201 TG respectively. This year, the range has been refreshed again. The TG appellations have been removed from the M 88 and M 201, and beyerdynamic are now promoting their classic designs as the M Series. The aim is to shift the focus back onto the timeless

quality of these designs, and in particular, their heritage as studio tools. To celebrate the relaunch of the M Series, I was invited to Heilbronn to see how these classic mics are assembled — a process that, in some cases, has not changed for 60 years or more.

All Systems Go

As Production Director Thorsten Bender explains, it is a key selling point of most beyerdynamic products that they are made in Germany, and for this reason, "system production" for professional headphones and the majority of microphones happens entirely within the plant. What this means, in effect, is that every aspect of manufacturing that makes a difference to the sound or the performance of a product takes place in Heilbronn, from the component level up. Of the 350 or so people employed at Beyer's headquarters, 230 are directly involved in production, and the breadth of operations here is remarkable.

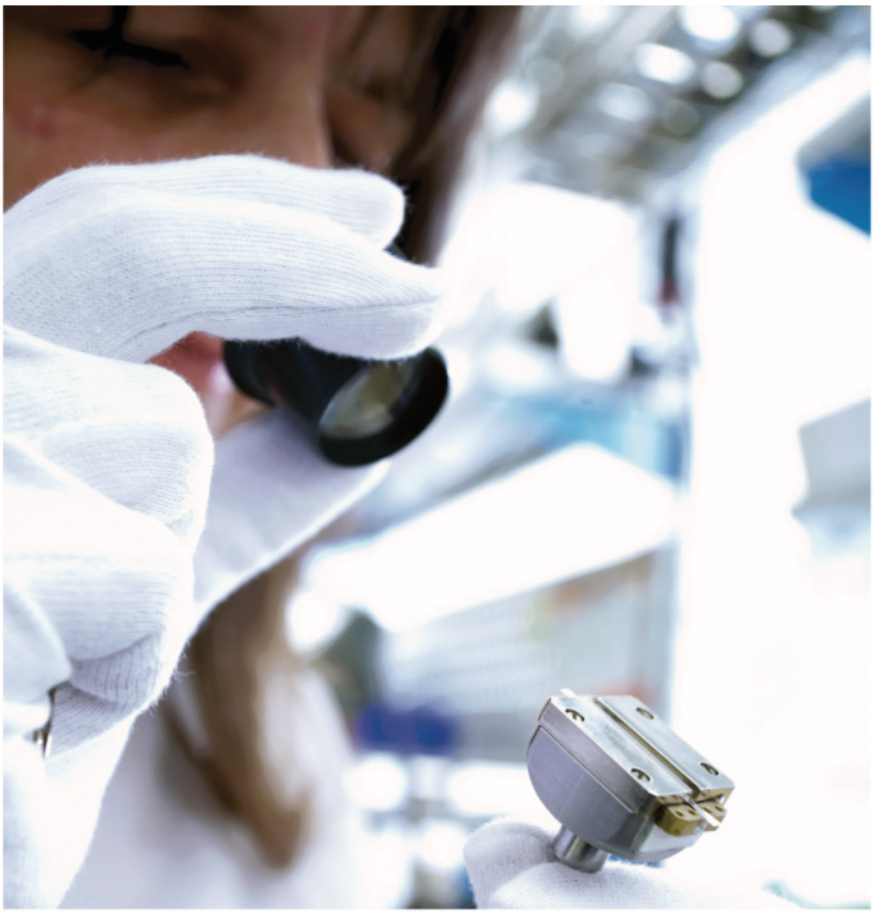
One thing that's very striking about the beyerdynamic factory, and indeed those of other microphone manufacturers I've visited, is that the people building the products are all women. In fact, Thorsten says, there's not a single man among the hundreds of workers employed in the manufacture of beyerdynamic mics or headphones. On joining the company four years ago, he was invited to try his hand himself and was politely told "You'll make a better manager than a worker!", and his experience is apparently typical: very few men, he says, have the fine motor skills required to reliably assemble small, complex and highly delicate items to an incredible degree of precision.

Even those who do have the requisite natural gifts require a fair amount of training and experience to get up to speed. A typical task might take four to six weeks to learn to perform competently, and there may be many such tasks in the production of a single item. To make production flexible enough, all factory operatives need to know how to do all of these tasks, so full training can take a year or more. The most specialised and most highly skilled workers of all are the five women who handle ribbon microphone production.

Vintage Modern

Introduced in 1957, the beyerdynamic M 160 ribbon microphone is the oldest product in the company's current line-up. When it was developed, it incorporated several technological breakthroughs that are still unique today. By stacking two short ribbons on top of each other, beyerdynamic were able to make the ribbon motor very small without sacrificing sensitivity. A unique corrugation process allowed the ribbons to move pistonically in response to vibration, while the addition of a plastic waveguide further increased sensitivity and improved the high-frequency response. And by connecting the back of the ribbon motor to an acoustic labyrinth, beyerdynamic were able to modify its native figure-8 polar pattern to give a hypercardioid response.

The M 160 and its figure-8 sibling, the M 130, remain in the beyerdynamic catalogue more than 60 years later, and are still produced almost entirely by hand, using the same toolings and techniques. The ribbons themselves, for example, are shaped and cut using an ingenious method I haven't seen elsewhere. A small sheet of aluminium foil is clamped into a guillotine-type device with flat jaws that form a template, stamping the distinctive »



A worker inspects the alignment and tensioning of the ribbon assembly in an M 160 motor.

To corrugate and shape the ribbon, a small sheet of foil is clamped into this jig, with the excess cut away using a razor blade.

There are dedicated test environments on the factory floor for some models, but ribbon mic performance needs to be checked in a full-sized anechoic chamber.

» corrugation pattern into the foil. A razor blade is then used to cut away the excess material, so when the jaws are unscrewed, the ribbon emerges precisely the right size and shape every time. Only one ribbon can be cut at once. Measuring perhaps 20mm by 2mm and impossibly thin, they have effectively no mass, so as Microphone Product Manager Milena Baur points out, ribbon motor assembly is definitely a job that can't be done with the window open!

The M 160 motor is particularly complex, and is put together entirely by hand. Once the frame is bolted together, the first of the ribbons can be glued in place. This is done by eye, to an almost unbelievable level of precision — the clearance between the edges of the ribbon and the magnets is a 10th of a millimetre. Not only must the ribbon sit exactly centrally in the gap, but it must also be tensioned correctly, both of which are achieved by sliding the ends of the ribbon around whilst the glue is still wet. Once this is done and the glue is dry, a metal spacer is added at either end, and the second ribbon is glued in place, the gap between them being less than a millimetre.

Every other aspect of ribbon-mic production is carried out by hand to a similar level of detail, exactly as it has been since 1957. The baskets, for example, are made in-house from layers of steel mesh — three for the M 130, four for the M 160 — optimised to protect the ribbon from wind blasts without colouring the sound. The M 160's tubular body houses the acoustic labyrinth that helps to shape the polar pattern, which is an equally complex design. Construction of the M 130 is even more involved, thanks to the large spring-like structure that provides the magnetic flux return path around the motor. Every individual M 160 or M 130 is the result of several hours' work, and given that this takes place in a high-wage economy such as Germany's, it's surprising that these mics are still as affordable as they are. The unique design and skills involved in their manufacture also mean that, until now at least, no-one has been able successfully to copy them.

M & M

The beyerdynamic M Series also includes a measurement microphone, the MM 1; two small-diaphragm capacitor mics, »

Even processes such as fitting the protective gauze and name bands to the body of the MC mics are carried out by hand.

Making Headphones

There are two main strands to beyerdynamic's business: headphones and microphones. In terms of volume, headphone production is by far the larger of the two, and the factory is capable of turning out several thousand drivers for a particular model in a single shift. For this reason, a certain amount of automation is involved. Nevertheless, there is still a surprising amount of skilled work done by hand, especially on models such as the DT 990 PRO.

beyerdynamic's commitment to German manufacturing is wholehearted and actually extends to the complex machinery used in "system production", much of which was developed in-house. The first stage in producing a headphone driver, for example, requires a precise number of turns of wire to be wound around a polished steel former and glued together to form a voice coil. Varying the gauge of the wire and the number of turns changes the impedance of the resulting headphone driver, and the machines can be set up to produce a number of different versions. Hardest to manufacture is the 600Ω driver, which uses wire thinner than a human hair.

Other custom machines cut and form the diaphragms from different types of plastic foil, and 'cook' them to achieve the desired level of rigidity. But on models such as the DT 990 PRO, it's down to human beings to slide the voice coils onto the diaphragms and glue them in place, with the impossibly fine wires extending at exactly the right angles. The glue is applied using a paintbrush; amazingly, says Thorsten Bender, tests have shown that Beyer's assembly workers are able to do this with a precision and consistency that can't be bettered by automation.

Once the diaphragm and voice coil are mated with a magnet assembly, the first of several stages of testing can begin. Each driver is tested for various qualities such as frequency, amplitude and phase response, and once assembled, each pair of headphones is also tested. The idea is to minimise waste by ensuring that defective parts are weeded out before they can reach the later stages of production, so if there's something wrong with an individual driver, it won't lead to a complete pair of headphones having to be thrown away or broken up. Also, faults that do occur are often down to imperfections in the acoustic material used to form baffles within the headphones, so it's often the case that a driver that tests faulty can be rescued by replacing the baffle.

The whole system is highly data-driven. Major elements such as driver assemblies and PCBs have their own individual QR codes, with test results being entered automatically into a database. This process has helped bring down rejection rates from 10 percent to below 3 percent. That's no mean feat when you consider that tolerances are set so low that there's no need for pair matching between the left and right drivers in a set of headphones.

SOUND ON SOUND

AWARDS

The Best New Products Of The Year, chosen by the readers of *Sound On Sound*

Voting is now open for the 14th annual SOS Awards and continues throughout the rest of October to the end of November 2023 at www.sosawards.com. The results will then be compiled, ready for announcement during January 2024.

Each category consists of a shortlist of nominations, chosen by the *SOS* editorial team, and we'd like you to tell us what you think are the outstanding products in each of the groups. As always, you are not required to vote in every category — if you don't have any strong opinions on some of the product groups, there's no need to vote for anything in those categories.

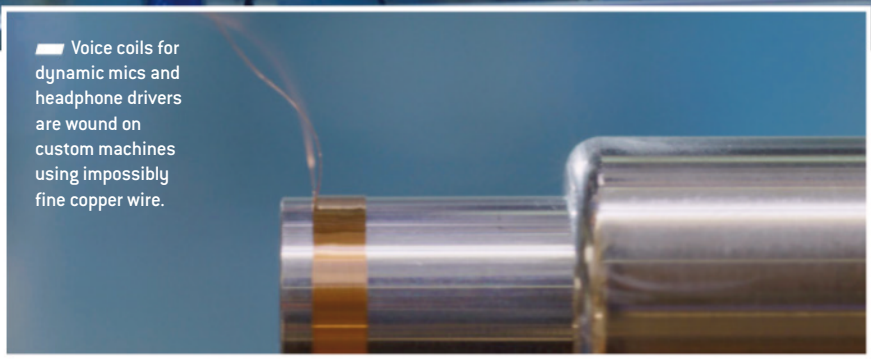
The categories are:

Audio Interface
DAW
Effects & Processing Hardware
Guitar & Bass Technology
Software Plug-in
Music Software
Performance Controller
Keyboard & Synth
Drum Machine, Sampler & Sequencer
Microphone
Mixer & Mixing Controller
Monitor
Hardware Recorder
Mic Preamp
Software Instrument
Studio Headphones & IEMs
Live Sound Product

To be nominated for an SOS Award a product has to have been on sale, or tested and reviewed by us, in the 12 months prior to the voting period. This year's nominations can be viewed at the URL below until the end of November 2023, and we very much look forward to seeing your choices for all the best new products of the last year in music technology and recording.

www.sosawards.com

(voting closes 30th November 2023)



Voice coils for dynamic mics and headphone drivers are wound on custom machines using impossibly fine copper wire.

» the MC 930 and MC 950; and two very well-known moving-coil mics, the M 88 and M 201. Sadly, the MC 840 large-diaphragm mic was discontinued a few years ago.

All of these mics boast a lengthy history, with the M 88 and M 201 dating back to the '60s and '70s respectively. And they all share with the ribbon models the fact of being very largely handmade. Both capacitor mics use the same capsule: a conventional, but not simple, true capacitor design that is built up by hand, even down to the process of attaching the diaphragm. To give you an idea of the fine tolerances at play in microphone construction, the only difference between the cardioid MC 930 and supercardioid MC 950 is the inclusion or otherwise of a single spacing washer — this alone is enough to modify the polar pattern.

Thorsten Bender is beyerdynamic's Production Director.

Being phantom-powered capacitor mics, the MC 930 and 950 have active circuitry, and this is one aspect of production that is highly automated. Printed circuit boards for these and many other beyerdynamic products are produced on high-tech machines: a pick-and-place robot populates the boards with surface-mount components before wave soldering fixes them in place. Even in this very modern, automated environment, though, there's still a role for human hands and eyes. Larger components such as the MC mics' switches must be attached manually, and as well as electrical testing, each completed PCB goes through a rigorous checking process whereby a scanning machine flags up visual discrepancies between the unit being tested and a reference sample, and a worker judges whether these are significant enough to merit rejection.

Keep On Moving

Like the ribbon mics, the M 88 and M 201 are still made in exactly the same way as they always have been; and, once again, it's a highly skilled, labour-intensive process. Although both are moving-coil dynamic mics, the diaphragm of the M 88 is more than twice as wide as that of the M 201, and the two microphones are therefore more different in sound and application than their specs might suggest. As in the older headphone designs (see box), the diaphragms are formed from a polyester film called Hostaphan which, fortunately, is still available today. Once cut and formed into the appropriate 3D shape, this is heat-treated to set it, whereupon the voice coil can be attached — by hand, naturally.

The entire transducer assembly is quite complex, incorporating the acoustic labyrinth that controls the polar pattern as well as a second humbucking coil and an inductor to minimise noise and

interference pick-up. One of the most crucial and challenging moments in the assembly process is that of attaching the diaphragm, with its voice coil glued in place, to its mount. The voice coil must sit precisely in the middle of a narrow circular gap, so that it can move freely yet be close enough to the surrounding magnet to generate a healthy signal level.

The use of strong magnets within these mics introduces production difficulties of its own. Because the metal must be magnetised prior to construction, there's a constant risk of ferric dust and debris being attracted, and it only takes a tiny particle of iron to compromise performance. Thorsten Bender says that Beyer's designers have experimented with magnetising the metal in situ after construction, but so far they haven't been able to achieve satisfactory results.

A stringent testing regime is fundamental to quality control. For each M Series model, beyerdynamic have a library of 'golden samples' from different eras that serve as references; these themselves are tested at regular

Microphone Product Manager Milena Baur.

intervals and, if necessary, recalibrated or replaced with new ones. Each and every production mic is measured and compared against these references to give the individual plots that are supplied to buyers, and many individual components and sub-assemblies are tested before final assembly. It's all part of a culture that values the company heritage without being enslaved by it. beyerdynamic haven't stopped innovating, most recently in the consumer sphere with their high-end Xelento in-ears, which use a single full-range dynamic driver rather than the more usual balanced armature design. But they know the value of their established designs, and they are keen to reiterate that any changes they are making in relaunching the M Series really are purely cosmetic.

The microphone market has changed out of all recognition since the M Series models were first introduced, and as Thorsten Bender acknowledges, good quality microphones have become a lot more affordable. But what only a manufacturer such as beyerdynamic can offer is long-term consistency: the knowledge that if you buy an M 88 or an M 160 today, it'll sound exactly like the one you used in 1978. "You can buy a cheap microphone from a large retailer and it can sound great," he cautions. "But then you buy what is advertised as the exact same microphone two years later and it might be completely different!"

SAMPLE LIBRARIES

Bunker Samples

Bunker Perc

Kontakt Instrument

★★★★

Bunker Samples return with their new library Bunker Perc, recorded in the bomb shelter that gives the company their name. This is certainly not your standard percussion library, and that's what makes it interesting. With Bunker Perc, the Bunker (built at the end of the Second World War, and later repurposed as a recording studio) becomes the instrument.

The library requires the full version of Kontakt (6.7.1 or later), and includes a wide variety of creative, intense, industrial sounds created by hitting, slamming and otherwise musically interacting with the infrastructure of the bunker, with samples including door slams, handle clicks, hydraulic risers and air filter hits.

These sounds are well recorded (with a pair of AKG C414s) and consistently edited. While the raw recordings are available, they are also warped into a collection of sound-designed hits, which make up the bulk of the library. The warping is tasteful and doesn't obscure the organic feeling of the original recordings. The library has no recorded dynamic layers, however this is not particularly noticeable. volume and EQ are used to simulate dynamics and this works very well. Each sound has an appropriate number of round robins; the high hits have more than the low booms, for example.

Speaking of low booms, thunderous sounds on offer here could easily find their place in some trailer music, while the impacts provide a variety of reverberant stabs which wouldn't feel out of place in a horror film. The low hits could replace (or work in conjunction with) an orchestral bass drum, just as some of the mid hits nicely imitate tom and snare drum sounds. The high hits provide some snappy, ticky sounds that would work well in an action sequence, or could be used for quick rolls to build into lower hits. It would be nice to have the option of two-handed mapping, to aid in performing rolls. Using the reversed patches allows you to trigger each sound backwards, and using the time-stretch function, the lengths of these sounds can be tailored for your tempo. They work well to build into the hits (which, in the reverse patches, can be triggered on release).

Each sound can be affected independently. Among other parameters, you can control the amount of punch, shape the EQ, and position each sound in a different place in the virtual room without affecting any other sounds. However, it would be good to also have the option to control these parameters globally for all sounds.

For fans of Stomp, or composers and producers looking for a heavy, in-your-face, angry percussion sound, Bunker Perc excels. It is an unconventional percussion ensemble that can spice up your orchestral percussion section. *Thomas Field*

$49

www.bunkersamples.com

Strezov Sampling

Jade Evolutions

Kontakt Instrument

★★★★★

Jade Evolutions can be regarded as a partner product to Strezov's existing Jade Orchestra, but with all-new content and a substantially different instrument construct. Evolutions is less about multisamples, round robins and legato modes, and far more about texture and timbral layers. It weighs in at a substantial 21GB.

Residing within the NI ecosphere, this Kontakt instrument sounds reassuringly expensive and authentic. The main thrust is provided by four sampled elements, which are loaded in a familiar 'four-corners-of-a-square' arrangement, with an X-Y pad providing fading/mixing between each element. The samples are loaded through the Zones page, before a number of additional parameters may be applied, all at the individual sample level. This includes ADSR envelope, high- and low-pass resonant filtering, EQ, distortion, compression and reverb/delay effects. It's a very comprehensive arrangement, which easily allows partials to come and go over time, exactly as one might expect in a cinematic suite of this kind. All of this, before you even attend to the manual control of the X/Y pad, which is also open to hardware control via MIDI CC.

Jade Evolutions oozes sheer quality; while the resultant output is hugely impressive, the individually constructed samples are stunning, and categorised to make it easy to locate a particular sample colour. All of the samples originate from East Asia, with a blend of strings, percussion, winds, vocals and chants, allowing for the conjuring of inspirational backdrops. Editing is easily navigable, with a healthy number of included factory patches, which may be used in original form or edited at will.

While the World nature of Jade Evolutions absorbs the centre of the instrument, there are two further sample channels which may be added to further bolster a given patch. Described as Synth Overlay A & B, these also provide pre-prepared samples, drawn from a collection of synths. This range is similarly substantial, with samples that range from basic saws to LFO-generated patterns and occasional blips. This could certainly provide an entire patch section of its own, but at the least, it provides a perfectly integrated process for bolstering a given harmony, which can be useful if your World elements are providing swathes of flowery colour.

Jade Evolutions occupies a place in the market which could be considered cinematic, but it could almost certainly have its uses elsewhere, as a World library. Used alongside Jade Orchestra for lead-line colour or melodic detail, leaving Evolutions to provide the perfect Asian-inspired backdrop, would undoubtedly work well. It's impressive, nuanced and colourful, much like the part of the world that has inspired the collection. *Dave Gale*

€249

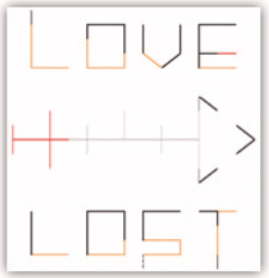
www.strezov-sampling.com

Sam Thomas Music

The Sound Of Love Lost

Kontakt Instrument

★★★★★



Sam Thomas is a composer, producer and session musician who describes his debut solo plug-in as being somewhere between a synth, guitar and organ. This is a sound that he has been honing as a guitarist over the last two decades, although it has recently been used in a track he released with Isamaya Ffrench called 'Love Lost', from which this library takes its name.

Compatible with the free and full versions of Kontakt 6.8.0 and above, Lost Love is just 3.57GB with a very simple-looking interface. The plug-in is intuitive enough, but also includes helpful mouse-over instructions. The sound was created using a Boss GT-8 pedal to add the synth, pitch and wah elements, then passed through an array of amps and effects. There are 13 samples per note across the full 88 keys and each has 20 seconds of natural sustain, achieved with just the force of the sound itself in the lower register and by utilising an Ebow and sustain pedal to achieve the same sustain in the higher register. The sound was completed by Sam and engineer Anni Abigail by adding EQ, saturation and stereo width during the mastering process.

There are two elements to the sound — the first is a deep, muted growl, filled with harmonics and ideal for bass and lead lines (Love) and the second, a lighter sound that includes a wah, which can be softened to create an ambient pad effect (Lost). These two elements can be used individually or blended to great effect. There is also the option to switch each to mono or add a low-pass filter to the Love element.

Below this is an attack fader with a range of 0-10, 0 being the slowest setting. Engage the Reverse button to slow the attack time down even further. Holding the notes down produces a very glitchy, organic sound that's verging on feedback.

The Space section of the plug-in brings in a mixture of reverb and delay with the option to adjust the tail, allowing for some very large, ambient effects. For the Wah section, each note was sampled 11 times. The wah notches can be triggered using either a slider, mod wheel, velocity or keyswitching. The effect can also be locked so that it's unaffected by velocity. Another addition to the plug-in, again taken from the song, is the option to blend two tunings, A=440Hz and A=432Hz. The slider simultaneously drops the pitch and inverts the stereo field.

Although this whole library is based around one GT-8 sound, it's very versatile due to the separate elements and extensive processing. If you're looking for a really thick, organic, guitar-based sound with plenty of harmonics then this is a great addition to your collection.

Atheen Spencer

£37

www.samthomasmusic.com

Spitfire Audio

Jupiter By Trevor Horn

Plug-in Instrument

★★★★½

UK producer Trevor Horn was a dominant force in 1980s pop, creating a string of massive hits as long as a Donald Trump charge sheet: 'Video Killed The Radio Star', Frankie Goes to Hollywood's 'Relax' and 'Two Tribes', ABC's *Lexicon Of Love* and Grace Jones' *Slave To The Rhythm*, to name but a few. Horn's productions are underpinned by a solid musicality, bold orchestrations, a deep familiarity with studio equipment and a ruthless focus on achieving the perfect sound for each overdub. All of which is good news for buyers of Jupiter By Trevor Horn, a collaboration with Spitfire Audio which gives users instant access to the maestro's sound world.

The library (45.59GB downloaded) contains 208 of Horn's multisampled sounds divided into Drums, Bass, Piano, Synths and FX. From these materials Spitfire fashioned 300 presets housed in their dedicated Solar plug-in, which utilises the company's twin-bay eDNA interface. The presets mostly feature a pair of layered sounds with a built-in Oscillate Mixer enabling you to crossfade and auto-pan between them.

Each sound is presented in its raw state along with four burned-in 'warps', which include gated period reverbs, a mangled, worn-out VHS tape noise and modulation effects created on Roland's Dimension-D and -C chorus units. The chorusing sounds overdone to me, but that's not a problem: you can create your own custom treatments by applying the interface's extensive effects to the unprocessed core samples.

Jupiter's ultra-punchy drum and percussion hits were sampled from classic drum machines (Oberheim DMX, a modified Linn 9000, Akai MPC2000XL, E-mu SP-1200, Roland TR-808) and processed with tape, varispeed and outboard effects. Though many are drenched in 1980s-style gated reverb, some less identifiable dry, crunchy hits would work well for contemporary productions. Accompanying the drums are a superb set of bass guitars from Trevor Horn's personal collection and a very decent, eminently playable Bösendorfer grand piano.

The heart of the library is its synth sounds, captured from collectors' items such as a Roland Jupiter-8, Juno-106 and JX-8P, Oberheim Matrix, Novation Bass Station and Solina string synth. Spitfire's presets are colourful, imaginative and highly varied, ranging from beautiful soft pads to rich, mobile sound tapestries such as 'Epic Pad', which features some distinctly non-vintage orchestral strings. The FX samples (some of which have a mad sci-fi flavour) are equally ear-grabbing.

Pro users will likely want to isolate individual synth sounds for programming purposes, and that's where things get difficult. The synth sounds are not listed alphabetically, forcing you to scroll endlessly through 123 randomly-organised names to find the sound you want, while the space-wasting interface design introduces new levels of squinting to browser navigation. This takes nothing away from Jupiter's musical content, which should open up new paths of creativity for its users — welcome to the pleasuredome, indeed.

Dave Stewart

$129

www.spitfireaudio.com

[Audio examples of this month's libraries are available at www.soundonsound.com.]

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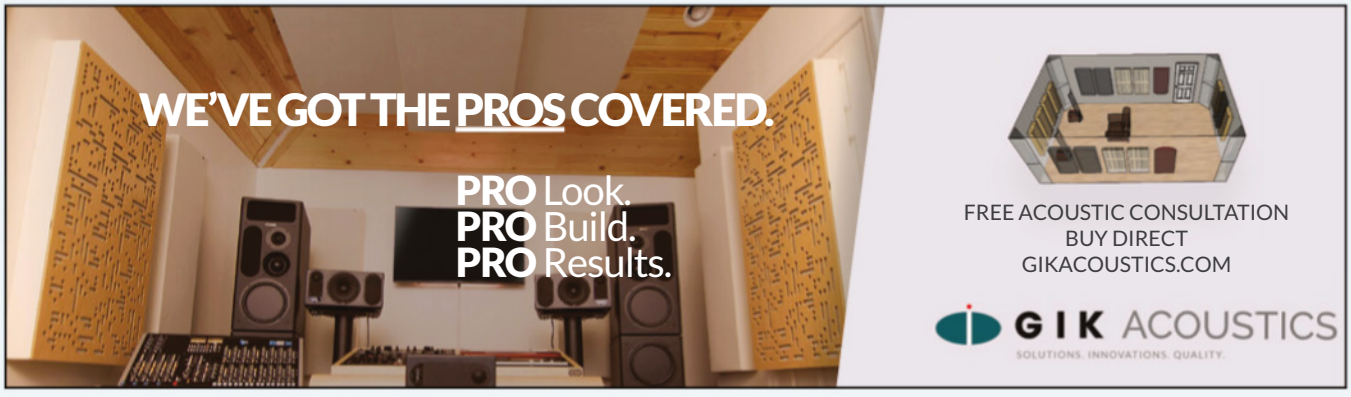

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SOS WHY I LOVE...

MY LES PAUL JUNIOR

SAM MURRAY

Working as both a session guitarist and music educator, I am often asked about whether I have a most-used or go-to guitar, and to be honest, the answer is no. Each guitar I have played on a session or a gig has served a different and context-specific purpose. But if you were to ask me which electric guitar I most pick up when I play for pleasure, not work, I do have an answer, absolutely and definitively. For me, that guitar will always be my Gibson Les Paul Junior.

I still remember the first time I heard one; it was on a classic rock compilation album, and it was a track by the band Mountain, featuring the late, great Leslie West on guitar. I had never heard such a visceral and raw tone, and was immediately obsessed with finding out what had been used to get that sound. By chance, shortly afterwards, a guitar magazine published an article on West that described his mastery of the Les Paul Junior guitar. That was it. I was sold; this was an instrument I needed to play. I must confess, it took a long while before I got my hands on one (a story for another day perhaps), but when I did, it certainly didn't disappoint.

Is it the perfect guitar? No, not by any means, if there is such a thing anyway. The wraparound bridge is not known to be the best; some of the necks have been described as feeling like half of a baseball bat; and to top things off, it only has one pickup. But it is precisely this lack of features that I love. This guitar does me no favours and makes me work harder than any other I have ever played. It also doesn't hide any of the nuances of my playing. I can absolutely see why Leslie West once described these simple, unadorned instruments as being like "a tree with a microphone".

Admittedly, in some higher-gain situations, I have had sound engineers and producers complain about the background (and, to be honest, sometimes foreground) hum that emanates from the guitar's single P90, but this is part of its charm. Not for me, your noiseless pickups. I like the hum. It means the game is afoot. It is a promise of things to come.

I am certainly not alone in my enthusiasm for these guitars; do a quick web search and you'll find amazing performances on the humble Junior from Steve Howe, Gary Moore, Keith Urban, Ritchie Sambora, Keith Richards, Phil X and many more. There are even some photos of a young Eddie Van Halen playing one; apparently it was one of his guitars of choice back when a fledgling Van Halen were playing Californian house parties in the mid-'70s.

I have played lots of guitars in my time, and they all do different things really well, but the one that I reach for when I feel tired or jaded and need to reignite my enthusiasm for all things guitar is my Les Paul Junior. In a world of numerous, perhaps more prestigious guitar choices, it might seem odd that I am so drawn to this scrappy little underdog of an instrument. Then again, perhaps that is precisely why I love it.

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